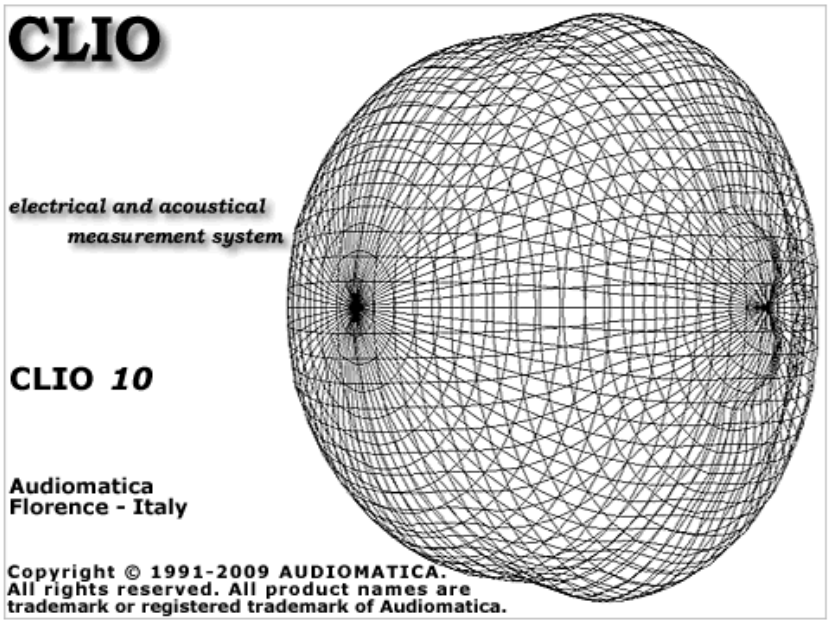
CLIO

ELECTRICAL & ACOUSTICAL TESTS



CLIO Software

Release 10
Version Standard

User's Manual

AUDIOMATICA

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1 INTRODUCTION

1.1 ABOUT THIS MANUAL

This User's Manual explains the CLIO system hardware and CLIO 10 software.

All software versions are covered, please note that CLIO 10 software is designed to operate in conjunction with the supplied PC hardware. If the hardware is absent or the serial numbers do not correspond then CLIO 10 will operate in **demo mode** only.

1.1.1 WHAT THIS USER MANUAL DOES COVER

The CLIO System is a complete electro-acoustic analyzer. There are thousands of books on many of the topics that CLIO handles as a measurement system. The simple definition of Frequency Response could be extended to a book itself. This User Manual is intended only as a **guide** to allow the user to quickly become efficient in using the CLIO system, its user interface, its hardware features and limits. Every topic is handled through real life examples with dozens of actual measurement being presented for clarity. It is therefore a HOW TO manual; WHY is left to the reader to explore through other literature and should be considered as essential reading. There is however reference [1], 'Testing Loudspeakers' by Joseph D'Appolito, which, in our opinion, is the perfect complement of what is covered here. Anyone who feels that WHY and HOW is strongly related should seriously consider buying this wonderful book.

1.2 GENERAL CONDITIONS AND WARRANTY

THANKS

Thank you for purchasing your CLIO system. We hope that your experiences using CLIO will be both productive and satisfying.

CUSTOMER SUPPORT

Audiomatica is committed to supporting the use of the CLIO system, and to that end, offers direct support to end users. Our users all around the world can contact us directly regarding technical problems, bug reports, or suggestions for future software enhancements. You can call, fax or write to us at:

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AUDIOMATICA ON-LINE

For any inquiry and to know the latest news about CLIO and other Audiomatica’s products we are on the Internet to help you:

AUDIOMATICA website: www.audiomatica.com
E-MAIL: info@audiomatica.com

AUDIOMATICA'S WARRANTY

Audiomatica warrants the CLIO system against physical defects for a period of one year following the original retail purchase of this product. In the first instance, please contact your local dealer in case of service needs. You can also contact us directly as outlined above, or refer to other qualified personnel.

WARNINGS AND LIMITATIONS OF LIABILITY

Audiomatica will not assume liability for damage or injury due to user servicing or misuse of our product. Audiomatica will not extend warranty coverage for damage of the CLIO system caused by misuse or physical damage. Audiomatica will not assume liability for the recovery of lost programs or data. The user must assume responsibility for the quality, performance and the fitness of Audiomatica software and hardware for use in professional production activities.

The CLIO SYSTEM, CLIOfw, CLIOQC and AUDIOMATICA are registered trademarks of Audiomatica SRL.

AUDIOMATICA REGISTRATION CARD
(EMAIL OR FAX TO US)

CLIO SERIAL NUMBER: _____

SOFTWARE VERSION: _____

PURCHASE DATE: _____

NAME: _____

JOB TITLE: _____

COMPANY: _____

ADDRESS: _____

ZIP OR POST CODE: _____

PHONE NUMBER: _____

FAX NUMBER: _____

E-MAIL: _____

2 THE CLIO SYSTEM

Depending on the hardware options that have been purchased, the CLIO system consists of the following components:

– The FW-01 firewire audio interface

– The MIC-01 or MIC-02 or MIC-03 (also Lite) microphones

– The PRE-01 microphone preamplifier

– The QCBox Model 5 power amplifier, switching and testing box

In the next few pages we will describe each component and give its respective technical specifications.

NOTE: Audiomatica reserves the right to modify the following specifications without notice.

2.1 THE FW-01 FIREWIRE AUDIO INTERFACE

The FW-01 Firewire Audio Interface sets new hardware precision standards for the CLIO System. The FW-01 unit has been designed to be a complete two channels professional A/D D/A audio front-end for your PC; it is connected to the computer by an IEEE-1394 link giving you maximum performances; it can be powered by the same link giving you maximum portability. The FW-01 performances (24 bit @ 192 kHz) represent state-of-the-art measurement capabilities for any audio device or acoustical test. The FW-01 is equipped with an instrument grade balanced input and output analog circuitry with an exceptionally wide range of output attenuation and input gain that allows an easy interface to the outer world; the input and output loopback capability with the internal ultra stable voltage reference permit a simple and precise calibration of the whole instrument. A switchable phantom power supply lets you directly connect an Audiomatica MIC-01, MIC-02 or MIC-03 as well as any other standard balanced microphone to any of the FW-01 input.

2.1.1 TECHNICAL SPECIFICATIONS

GENERATOR
Two channels 24 Bit sigma-delta D/A Converter
Frequency range: 1Hz-90kHz
Frequency accuracy: >0.01%
Frequency resolution: 0.01Hz
Output impedance: 660Ohm
Max output level (Sine):17dBu (5.5Vrms)
Attenuation: 0.1 dB steps to full mute
THD+Noise(Sine):0.008%
Digital out: SPDIF
ANALYZER
Two channels 24 bit sigma-delta A/D Converter
Input range: +40 ÷ -40dBV
Max input acceptance: +40dBV (283Vpp)
Input impedance: 128kOhm (5.6kOhm mic)
Phantom power supply: 24V
PC SYSTEM RESOURCES
One free IEEE1394 port
MISCELLANEOUS
Sampling frequencies: 192kHz, 96kHz and 48kHz.
Connections: analog 2 XLR combo in, 2 XLR plus 2 RCA out, 1 RCA digital out
Digital connection: 6-pin IEEE1394
Power supply: IEEE1394 or 12V DC
Dimensions :16(w)x17(d)x4(h)
Weight: 0.8 kg

2.2 THE MIC-01 MICROPHONE

The MIC-01 microphone is an electret measuring microphone that is particularly well suited to being used in conjunction with the other components of the CLIO system. It is furnished with its own stand adapter and a calibration chart reporting the individually measured sensitivity, all fitted in an elegant case. Its long and thin shape renders it ideal for anechoic measurements. Because its frequency response is very flat over the entire audio band, no particular correction is usually needed.

2.2.1 THE MIC-02 MICROPHONE

The MIC-02 microphone is functionally identical to MIC-01. It differs only in the fact that its length is 12 cm instead 25 cm. The MIC-02 is more practical to handle and to work with, and is ideal for measurements in a reverberant environment.

2.2.2 THE MIC-03 MICROPHONE

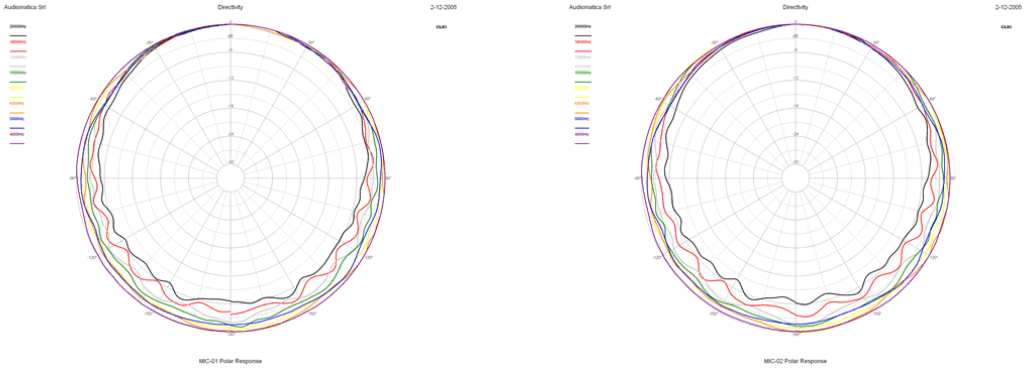
The MIC-03 microphone is functionally identical to MIC-01. It differs only in the fact that its length is 7 cm instead 25 cm. The MIC-03 is more convenient where space saving is a must.

2.2.3 TECHNICAL SPECIFICATIONS

MIC-01

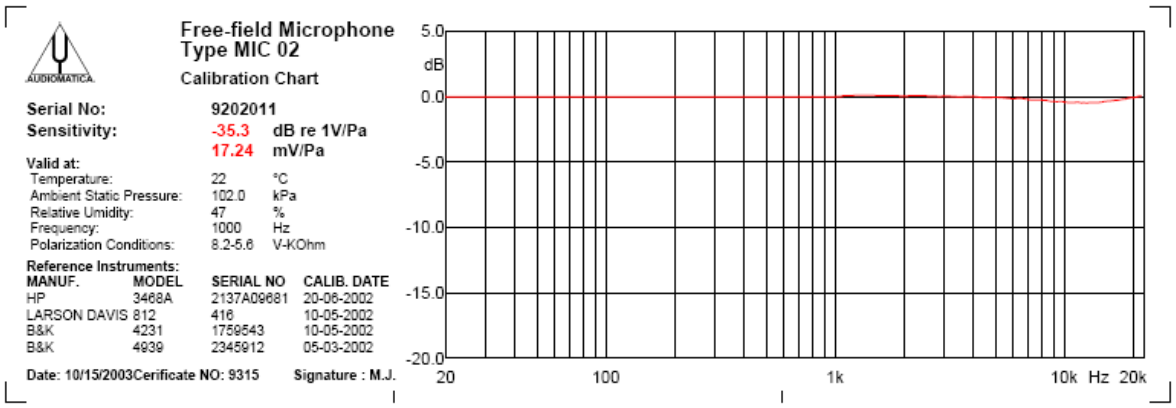
Type:	Condenser electret
Accuracy:	±1 dB, 20 Hz to 10 kHz
	±2 dB, 10 kHz to 20 kHz (direct field)
Maximum level:	130 dB SPL
Dimensions:	8 mm diameter, 25 cm long
Accessories:	wooden case, 2.7 m cable, stand adapter
MIC-02:	Same as MIC-01, but 12 cm long.
MIC-03:	Same as MIC-01, but 7 cm long.

Polar Response:



MIC-01-MIC-02-MIC-03

2.2.4 THE MIC-01 (OR MIC-02) FREQUENCY CALIBRATION DATA



The microphones MIC-01 and MIC-02 can be furnished with (or be submitted for) a frequency calibration certificate. This document, along with numerical data on floppy disk, is released by Audiomatica and specifies the frequency behavior of the single microphone under test. The file data can be used with the CLIO software (see 5.4.2).

2.2.5 THE MIC-01, MIC-02 or MIC-03 LITE MICROPHONE

In the **Lite** version of MIC-01, MIC-02 and MIC-03 the accessories (wooden case, 2.7 m cable and stand adapter) are not supplied. The microphone comes as in figure.

2.3 THE PRE-01 MICROPHONE PREAMPLIFIER

The microphone preamplifier PRE-01 has been designed to match Audiomatica's microphones MIC-01, MIC-02 and MIC-03. It is particularly useful when the microphone has to be operated far from the analyzer or when weighted measurements are needed. PRE-01 powers the microphone connected to its input with an 8.2V phantom supply and adds a selectable weighting filter (A or B or C); also available there is a 20 dB gain stage. The unit is operated with one standard 9V battery or with an external DC power supply.

2.3.1 TECHNICAL SPECIFICATION

Frequency response:	7Hz÷110kHz (-3dB)
Weighting filter:	A, B, C (IEC 651 - TYPE I)
Phantom power supply:	8.2V (5600 Ohm)
Gain:	0 & 20dB (INTERNAL SETTINGS)
Input impedance:	5600 Ohm
Output impedance:	100 Ohm
Maximum output voltage (@1kHz):	25 Vpp
THD (@1kHz):	0.01%
Input noise (@20dB gain):	7uV LIN, 5.3uV A
Drive capability:	±7mA
Batteries duration:	>24h (alkaline cell)
Size:	12.5(w)x19(d)x5(h)cm
Weight:	900g

2.3.2 USE OF THE PREAMPLIFIER

The MIC-01 or MIC-02 or MIC-03 microphone cable should be connected to the preamplifier input while the preamplifier output requires connection to the analyzer input. The unit is switched on with the POWER switch, while the TEST push-button controls the state of the unit. Correct operation of the unit is indicated by the led light being illuminated, if the LED fails to illuminate then either the batteries are low or the external power supply is not connected. The FILTER switch inserts the weighting filter. To choose the desired weighting filter type and to set the amplifier gain you have to modify the unit settings with the dip switch operated from the back panel.

NOTE: if the 20 dB gain stage is inserted the overall sensitivity (microphone + pre) is 10 times higher. For example if your microphone has a sensitivity of 17.1 mV/Pa and you amplify it by 20 dB then you get a sensitivity of 171 mV/Pa.

2.4 THE QCBOX MODEL 5 POWER AMPLIFIER, SWITCHING AND TESTING BOX

The QCBOX Model 5 power amplifier, switching and testing box is of invaluable help when configuring an automatic or manual quality control setup, or even in everyday laboratory use.

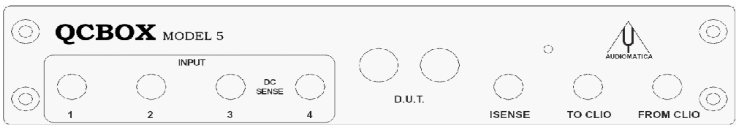
It can be configured, **under software control via USB**, to assist frequency response and impedance measurements or to perform DC measurements.

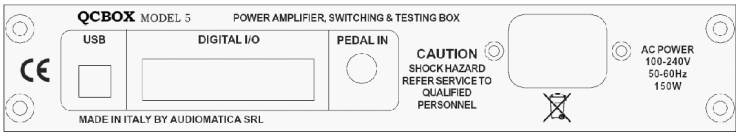
Between its features is the possibility of internal switching that permits the measurement of the impedance and frequency response of the loudspeaker connected to its output sockets **without changing the wiring to the speaker**; it is also possible to route one of four inputs for the response measurements; these input have powering for a microphone (0÷24V variable).

An internal ADC measures the DC current into the voice coil; an over current limiter is available to a predefined threshold. Thanks to an internal software controlled voltage generator the speaker can be driven with a superimposed DC voltage (±20V max), allowing for measurements of large signal T&S parameters. Two ADC converters with a ±2.5V and ±5V are available at inputs 3 and 4 to measure the displacement with a laser sensor or any other DC signal.

A dedicated output, ISENSE, allows impedance measurements in constant voltage mode as well as voice coil current distortion analysis.

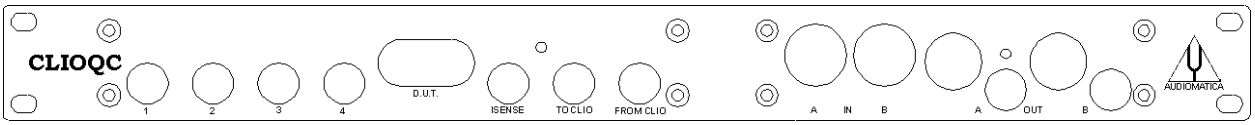
A 5 bit in - 6 bit out digital port is available to interface the QCBOX with external hardware or line automation. An ulterior dedicated input permits an external foot pedal switch to be connected and trigger QC operations.





19” RACK MOUNT ASSEMBLY

Using the Rack QC panel it is possible to assemble the QCBOX Model 5 together the FW-01 Audio Interface so that they can be mounted in a standard 19” rack frame.



2.4.1 TECHNICAL SPECIFICATIONS

Inputs:	Four line/microphone inputs with selectable phantom power supply (0÷24V variable) One TTL input for external trigger 5 digital lines
Outputs:	Isense 6 digital lines
Functions:	USB controlled internal switches for impedance and DC measurements
DC measuring:	Isense current ±2.25 A DC IN 3 ±2.5 V DC IN 4 ±5 V
Power output stage:	50W (8Ohm) with current sensing and overcurrent protection Possibility of superimposing a DC voltage (±20 V)
THD (@1 kHz):	0.004 %
Dimensions:	23(w)x23(d)x4(h)cm
Weight:	1.4kg
AC:	90÷240V

3 CLIO INSTALLATION

3.1 MINIMUM PC CONFIGURATION

The CLIO FW-01 firewire audio interface running the CLIO software can be installed in any personal computer with the following minimum system requirements:
– Pentium IV processor (suggested minimum 1GHz)
– One free IEEE-1394 port
– 256 MB RAM
– 1024x786 video adapter
– Microsoft Windows XP or Vista
– Adobe Acrobat Reader

DO NOT CONNECT THE FW-01 UNIT TO THE PC UNTIL REQUESTED!

If you are installing under:

- Windows XP go to section 3.2
- Windows Vista and 7 go to section 3.3

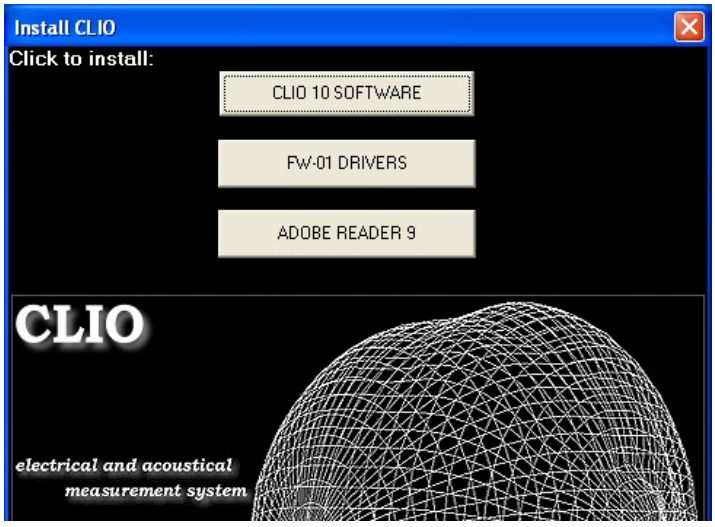
3.2 FW-01 DRIVERS INSTALLATION UNDER WINDOWS XP

The procedures described refer specifically (and are described with examples and figures) to the Windows XP Professional operating system, English version, they can be applied with only minor modifications and appropriate translations to all languages and to Windows XP Home.

To install the FW-01 drivers in your computer you should follow the instructions presented below:

1) Insert the CLIO 10 CD ROM in the computer.

2) Wait for autorun application or run "Clioinstall.exe".



3) Choose "FW-01 DRIVERS" to start installation.

WHEN PROMPTED CONNECT THE FW-01 UNIT!

To connect the FW-01 unit to your PC you need to do the following:

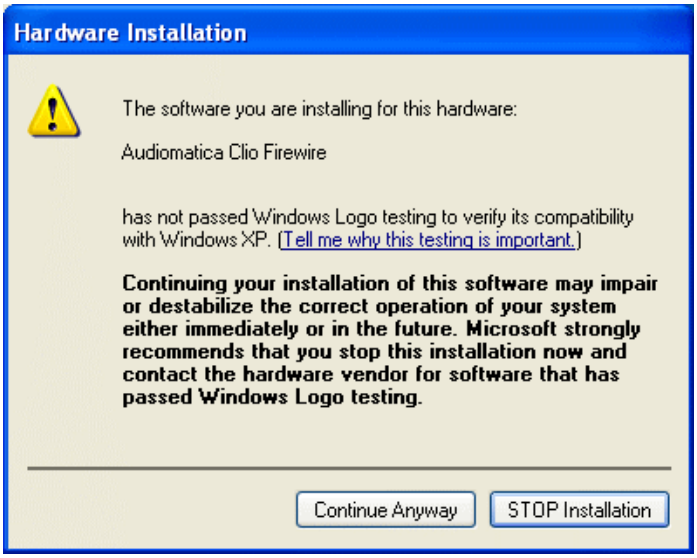
1) Locate an IEEE-1394 port on your PC. You may either use a standard 6-pin port (with or without power supply) or a standard 4-pin (small connector, without power supply) port.

2) If you use a 6-pin port use the supplied 6-pin-to-6-pin cable. If you use a 4-pin port please provide an IEEE 1394 6-pin-to-4-pin cable (often referred as i-Link).

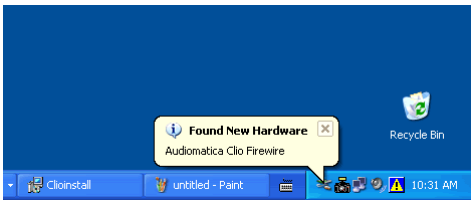
3) If you use a 6-pin port verify that it is capable of power supply.

4) If you use a 6-pin port without power supply or a 4-pin port you must also provide a 12V external power supply.

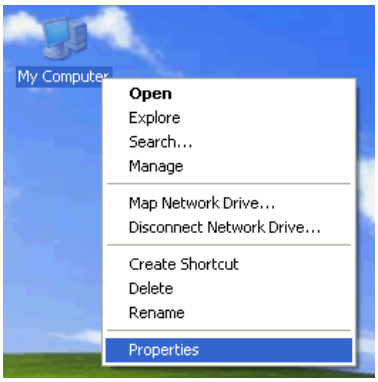
Ignore Microsoft's warning message about Digital Signature, answer 'Yes' to the prompt and reach the end of the wizard.

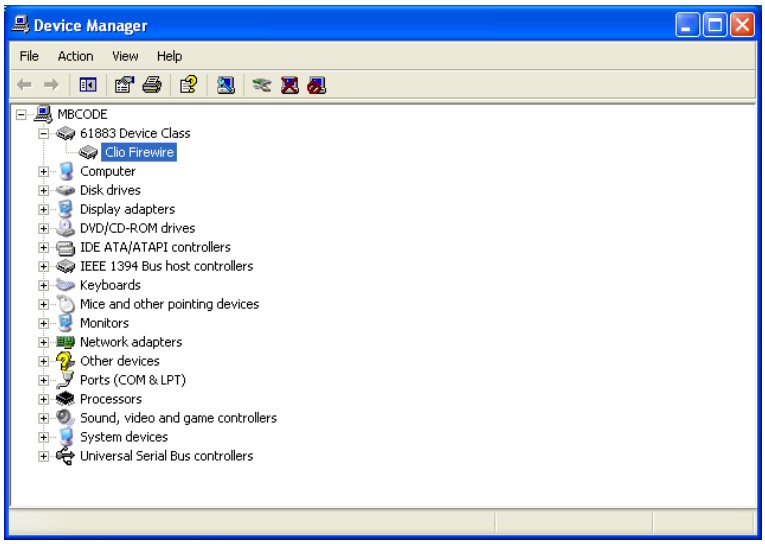

Hardware Installation
The software you are installing for this hardware:
Audiomatica Clio Firewire
has not passed Windows Logo testing to verify its compatibility with Windows XP. (Tell me why this testing is important.)
Continuing your installation of this software may impair or destabilize the correct operation of your system either immediately or in the future. Microsoft strongly recommends that you stop this installation now and contact the hardware vendor for software that has passed Windows Logo testing.
Continue Anyway
STOP Installation

You will then reach the end of the wizard.



Let's now verify the correct installation of the FW-01 driver. Click with the **right** mouse button on the 'My Computer' icon on the Windows desktop. Then click 'Properties', select the 'Hardware' tab and press the 'Device Manager' button as in the following figure.





Verify the presence of the 'Clio Firewire' driver under the 61883 device class.

Your driver installation was successful!

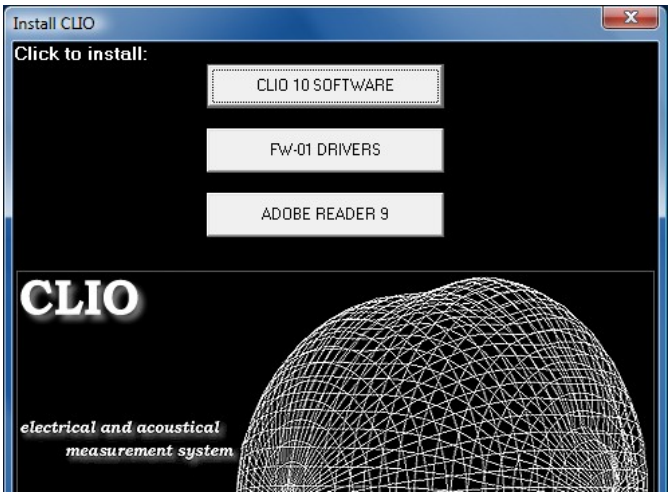
3.3 FW-01 DRIVERS INSTALLATION UNDER WINDOWS VISTA AND 7

The procedures described refer specifically (and are described with examples and figures) to the Windows 7 operating system, English version, they can be applied with only minor modifications and appropriate translations to all languages.

To install the FW-01 drivers in your computer you should follow the instructions presented below:

1) Insert the CLIO 10 CD ROM in the computer.

2) Wait for autorun application or run "Clioinstall.exe".



3) Choose "FW-01 DRIVERS" to start installation.

WHEN PROMPTED CONNECT THE FW-01 UNIT!

To connect the FW-01 unit to your PC you need to do the following:

1) Locate an IEEE-1394 port on your PC. You may either use a standard 6-pin port (with or without power supply) or a standard 4-pin (small connector, without power supply) port.

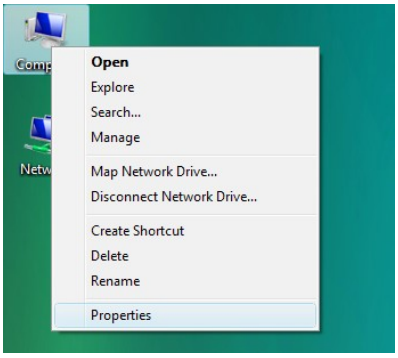
2) If you use a 6-pin port use the supplied 6-pin-to-6-pin cable. If you use a 4-pin port please provide an IEEE 1394 6-pin-to-4-pin cable (often referred as i-Link).

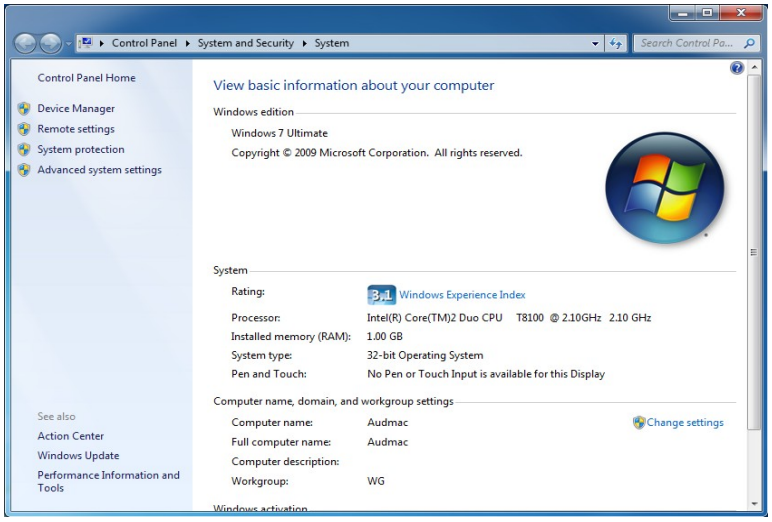
3) If you use a 6-pin port verify that it is capable of power supply.

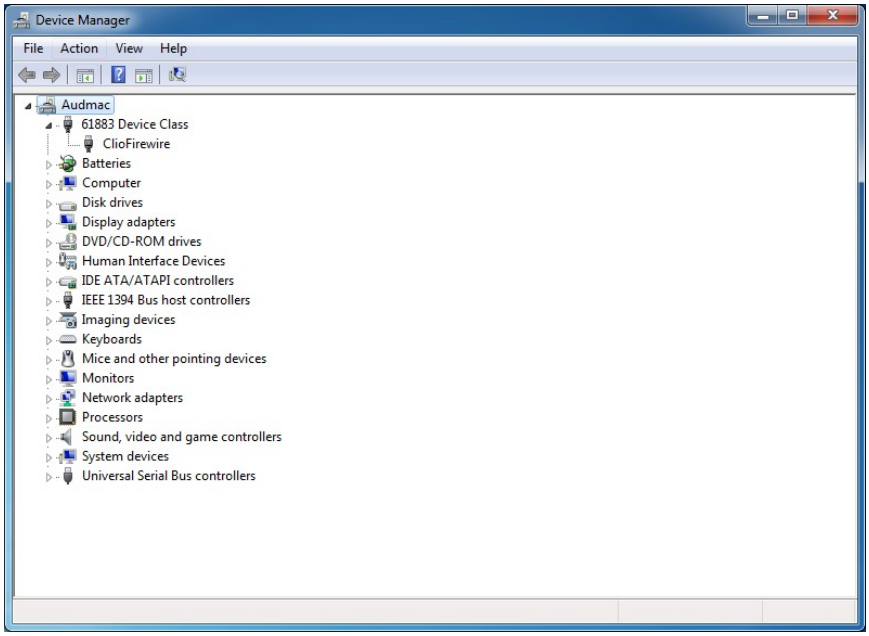
4) If you use a 6-pin port without power supply or a 4-pin port you must also provide a 12V external power supply.

You will then reach the end of the wizard.

Let's now verify the correct installation of the FW-01 driver. Click with the **right** mouse button on the 'My Computer' icon on the Windows desktop or on the Start Menu. Then click 'Properties', press the 'Device Manager' link as in the following figure.







Verify the presence of the 'Clio Firewire' driver under the 61883 device class.

Your driver installation was successful!

3.4 SOFTWARE INSTALLATION

This paragraph deals with **software installation.**

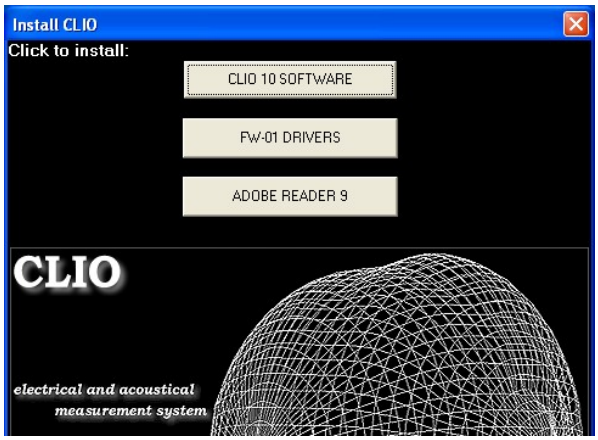
The CLIO software is provided either on its own CD-ROM or, in electronic format, as a single, self-extracting, executable file.

Be sure to have administrative rights when installing CLIO.

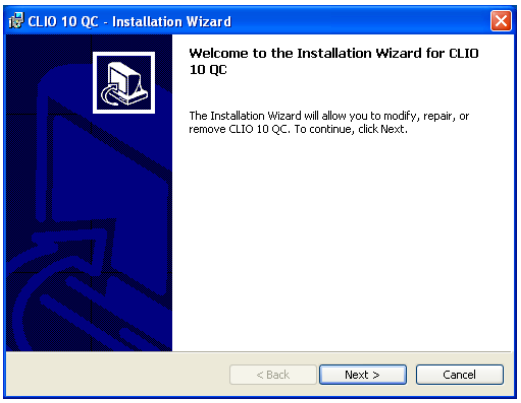
To install the CLIO 10 software in your computer you should follow the instructions presented below:

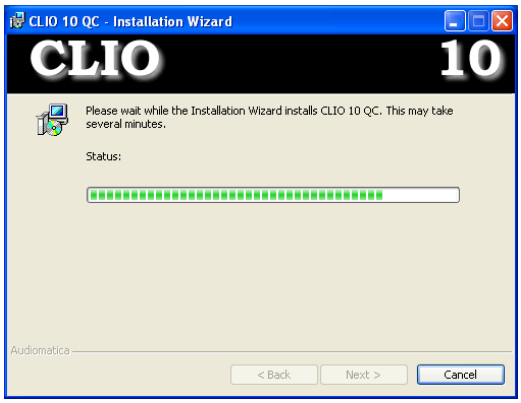
1) Insert the CLIO 10 CD ROM in the computer.

2) Wait for autorun application or run "Clioinstall.exe".



3) Choose "CLIO 10 SOFTWARE" to start installation.


CLIO 10 QC - Installation Wizard
Welcome to the Installation Wizard for CLIO 10 QC
The Installation Wizard will allow you to modify, repair, or remove CLIO 10 QC. To continue, click Next.
< Back
Next >
Cancel



The procedure is completely automatic and will only request you to accept the Software End User's License Agreement and input some information in order to correctly install CLIO 10; the software installer will also check your operating system version.

After successfully completing this procedure take note of the installation directory of CLIO (usually C:\Program Files\Audiomatica\CLIO 10).

3.5 THE 'CLIO BOX'

A few words about the FW-01 firewire audio interface.

Figure 3.26

This unit is needed to correctly interface analog signals to your PC; it is also important as it has an internal reference used to calibrate the system and also stores the system's serial number inside its internal EEPROM; Fig.3.27 shows where is located your CLIO system serial number.

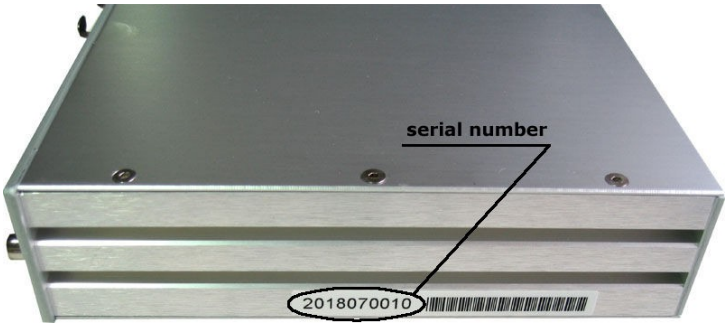


Figure 3.27

The serial number is very important and should be mentioned each time you get in contact with Audiomatica, either for technical support or for software upgrade.

When using your CLIO system you will normally use the FW-01 front connectors. As you'll become extremely familiar with this hardware unit we are going to give it a nickname: from now on we will call it '**the CLIO Box**'. Also the CLIO software refers to it with this nickname.

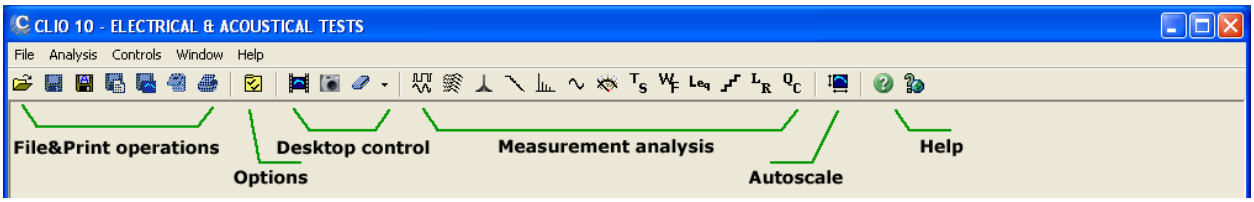
3.6 RUNNING CLIO FOR THE FIRST TIME

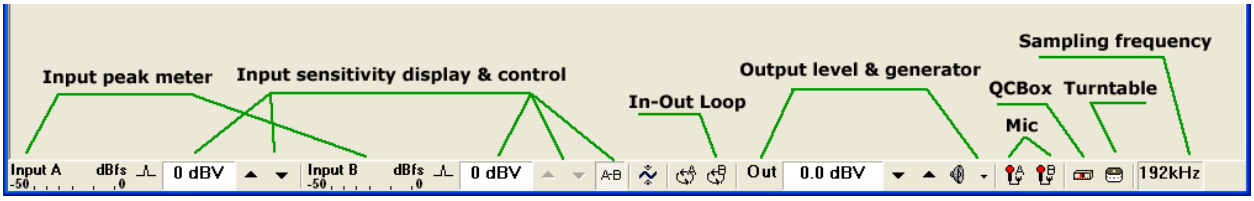
If you have completed the preceding installation procedure, you are ready to run CLIO!

The following steps will guide you through a complete verification of the system performance and operation.

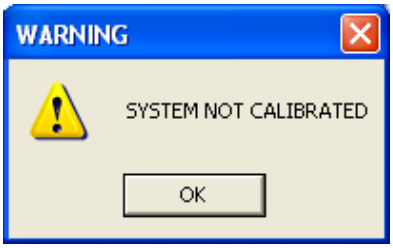
From the **Start Menu** choose **Programs**, then **CLIO 10** and click on the CLIO icon.

The program should start smoothly and present the main desktop.





If the the system is not calibrated, as the first time you run it, you will receive the following message.



Should CLIO display an error message take note of it and go to the troubleshooting section (3.9).

3.6.1 INITIAL TEST

Let's now execute our first test measurement - play and capture a 1kHz sinusoid.First of all click on the In-Out Loop button for channel A; in this way the CLIO Box connects output A with input A with an internal relay. This connection is very important as it lets you capture and analyze a signal generated by CLIO without the need for an external connecting cable.
Then click on the generator icon to play the 1kHz sinusoid (1031.25Hz to be exact; more on this later, it's the default signal). Then press the **F4** keystroke to invoke the Multi-Meter as in Fig.3.28.

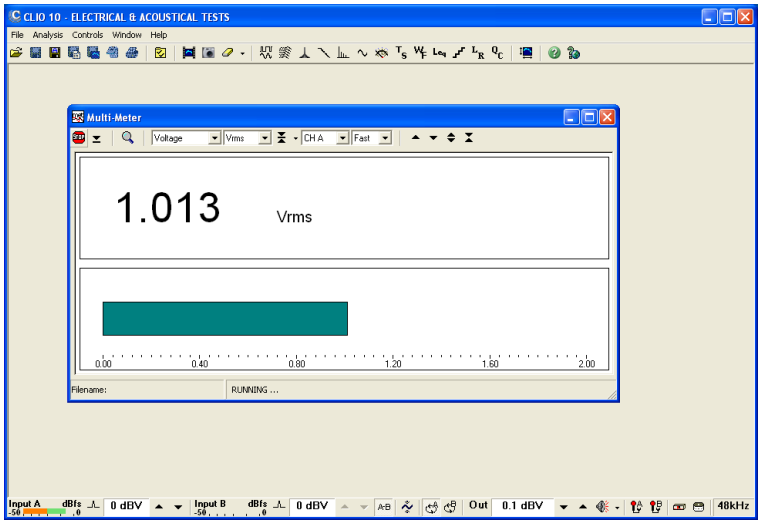
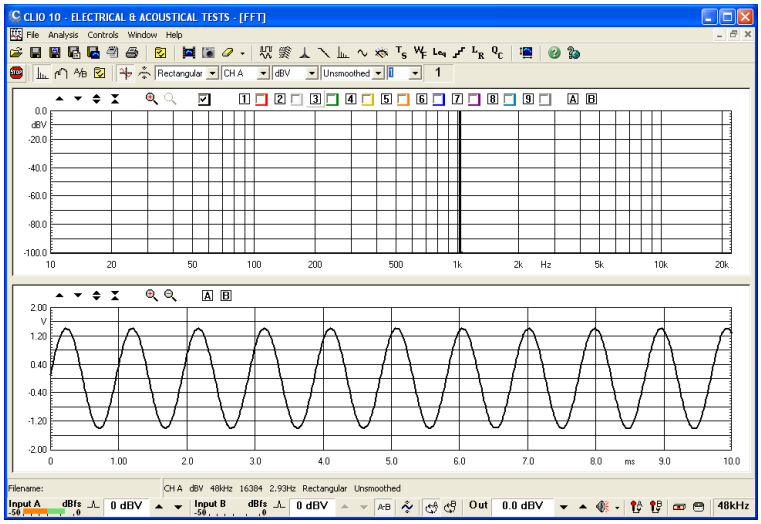


Figure 3.28

If everything is OK you should obtain a reading of circa 1V, variable between a minimum of 0.95V and a maximum of 1.05V, which is the mean output level of a sinusoidal signal when the system is not calibrated.

Now press the FFT button (or **CTRL-F**), then press the Oscilloscope button and finally the GoButton.



The result you should obtain is an FFT analysis of the 1kHz sinusoid (one spectral line @ 1kHz at 0dBV) and its time representation given by its oscillogram.

IMPORTANT NOTE: Only if these two initial tests gave correct results, as described, go to the following paragraph and execute the system calibration; if you are not able to obtain these results and they seem in any way corrupted **do not execute calibration** and contact technical support.

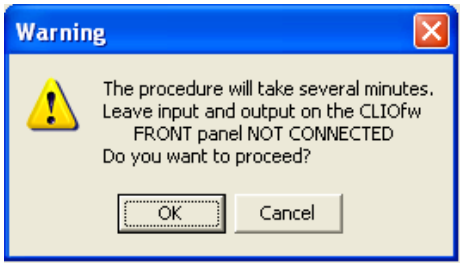
3.7 SYSTEM CALIBRATION

This section describes how to perform the system calibration.

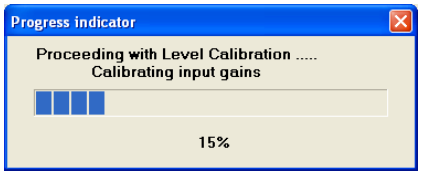
Be sure that, any time you perform a calibration, the system has warmed up for, at least 15-20 minutes.

Select **Calibration** from the **File** menu (5.6);

Leave the CLIO Box front plugs unconnected.



Answer **OK** to the initial prompt; this will run an automatic procedure that will last several minutes. The calibration procedure is completely automatic and several progress indicators will accompany all the executed measurements. At the end of it your CLIO system should be calibrated and ready to carry out measurements.



At the end of the calibration process it is always mandatory to verify the calibration itself; this is done by two simple measurements as described in the following section.

3.7.1 CALIBRATION VALIDATION

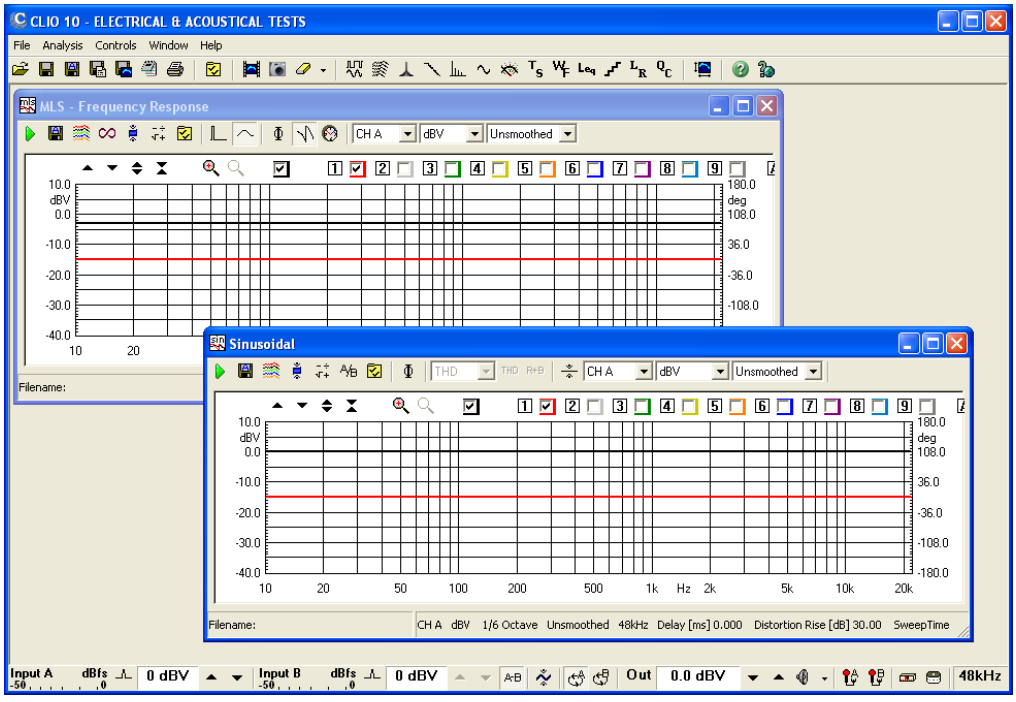


Figure 3.29

To verify the calibration first check that the generator output level is set to 0dBV (refer to 4.5.3 for details).

Press the channel A In-Out Loop button .

Then click on the MLS button to invoke the MLS control panel. Press the Go button to execute an MLS frequency response measurement; after about **1 second** you should obtain the desired result, a straight line (black) as in Fig.3.29. You can click on the graph and inspect the amplitude of the measured signal: you should obtain a reading around -3dBV, this is the correct output level of the MLS signal with the generator output set to 0dBV.

Now click on the Sinusoidal button to invoke the Sinusoidal control panel as in Fig.3.29. Press the Go button to execute a Sinusoidal frequency response measurement; after about **5 seconds** you should obtain the desired result, again a straight line (black) as in Fig.3.29. You can click on the graph and inspect the amplitude of the measured signal: you should obtain a reading around 0dBV.

To ensure a 100% correct calibration you also need to inspect the phase responses of both measurements. To do this press the phase button and verify that you obtain a straight line (red curves in Fig.3.29) the readings in this case should be around zero degrees in both cases.

As a final test repeat the 1kHz tone test described in 3.6.1. The expected result is shown in Fig.3.30.

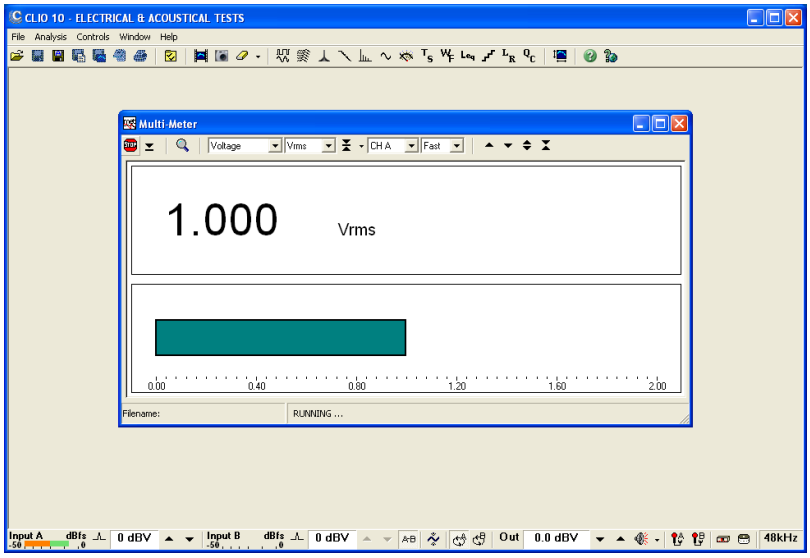


Figure 3.30

3.8 CLIO SERIAL NUMBER AND DEMO MODE

Each CLIO system has its own **serial number** which plays an important role since the CLIO software is hardware protected and relies on a correct serialization in order to run.

Refer to 3.5 to identify your system's serial number.

If the CLIO software doesn't find a CLIO Box with a correct serial number it gives a warning message and enters what is called **DEMO mode**; in this way it is possible to run CLIO in a PC where the CLIO hardware is not installed while still allowing you to perform post-processing and other off line jobs.

3.9 TROUBLESHOOTING CLIO INSTALLATION

To receive assistance please contact Audiomatica at info@audiomatica.com or connect to our website www.audiomatica.com.

4 CLIO BASICS

4.1 INTRODUCTION

This chapter gives you the basic information about CLIO and the related hardware and how to connect and operate it, while the following chapters explain in more detail the individual measurements available to users of CLIO. Chapter 5 deals with other general functionality of CLIO.

Here you will find information about:

- **Help**
- **Main desktop, toolbars and menu**
- **Shortcuts**
- **Generator, Input and Output, Microphone**
- **Amplifier & SwitchBox, Turntable**
- **Connections**

4.2 GETTING HELP

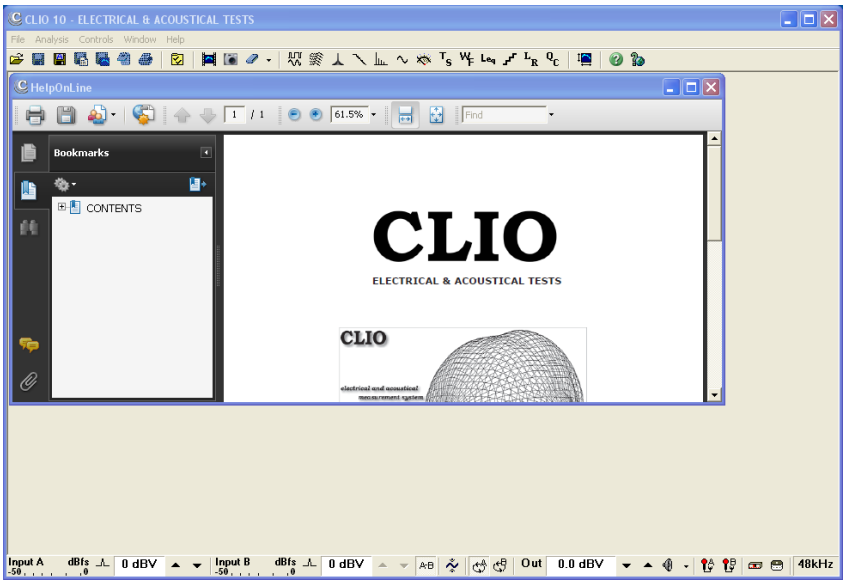


Figure 4.1 CLIO Help On-Line

To request the CLIO on-line help press F1. The on-line help screen (Fig. 4.1) should appear and the context-sensitive search should locate the page appropriate to the currently active menu, dialog or control.

*Note: in order for the CLIO help to work you should have **Adobe Acrobat Reader** installed on your system. The CLIO CD-ROM contains a correct version of this utility. Refer to Adobe (www.adobe.com) for any further information.*

The CLIO help can be invoked also from outside CLIO; to do this go to the Start Menu, then Programs, then CLIO and then click on 'CLIO Help'; in this way Acrobat will let you read and print this User Manual.

If you are not familiar with Acrobat, please spend some time to familiarize yourself with its capabilities, controls and navigation buttons.

Another way to obtain help is through the Help Menu (see 4.6.5) which gives you the possibility to view the on-line resources available in the Audiomatica and CLIO websites.

4.3 CLIO DESKTOP

The CLIO desktop presents itself as in Fig. 4.2 and gives you access to the **main menu**, the (upper) **main toolbar** and the (lower) **hardware controls toolbar**.

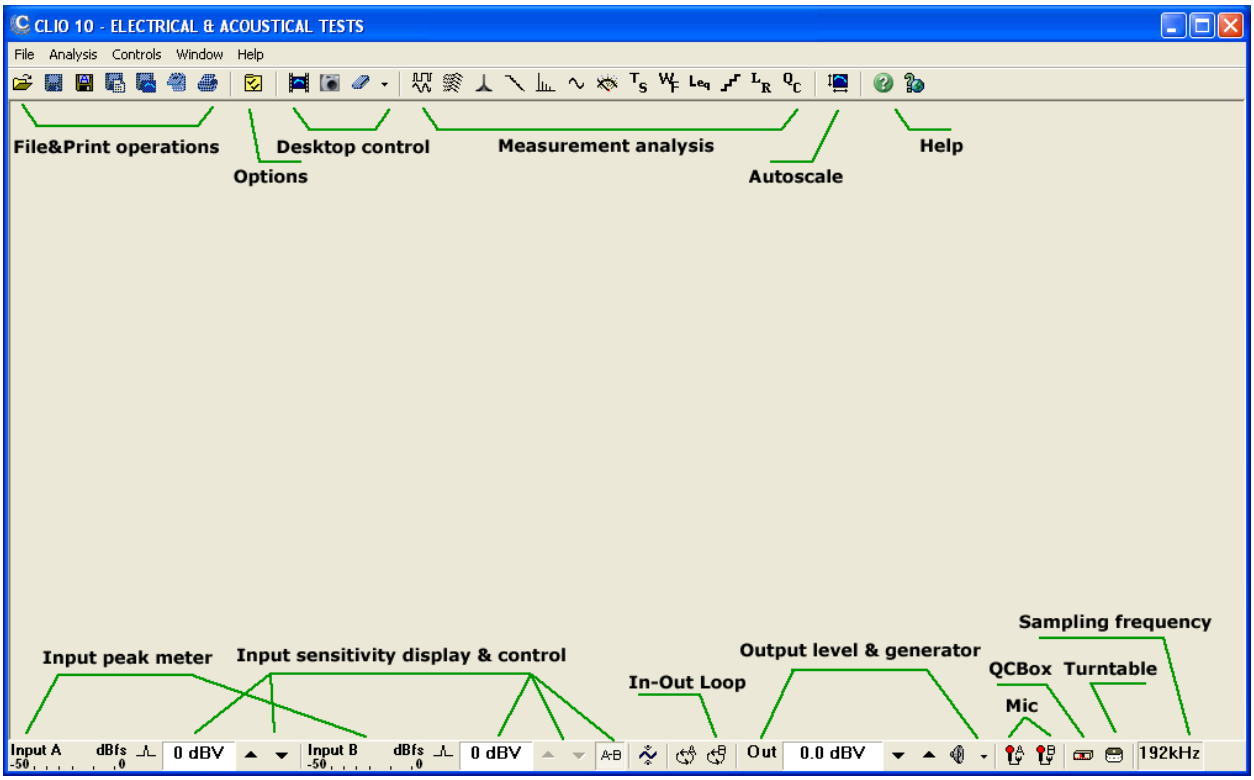


Figure 4.2 CLIO Desktop

Inside the main toolbar and the hardware controls toolbar you can locate several distinct functional areas as shown in the above figure. There now follows a description of all the controls inside the two toolbars. Refer to Section 4.8 for a detailed view inside the main menu.

4.4 MAIN TOOLBAR

Please refer to Chapter 5 for information about **File and Print functions**, **Options** and **Desktop control**.

4.4.1 MEASUREMENT ANALYSIS

By clicking on these toolbar buttons it is possible to interact and display each measurement control panel. Once the toolbar button is clicked the appropriate panel will be opened or reactivated. Any currently active panel will automatically be deactivated on activation of the new one.

The same functionality will be obtained with the relative **shortcuts** or by making a selection inside the Analysis Menu (see 4.6.2); a third way is to select a window through the Windows Menu (see 4.8.4).

Enters the MLS&LogChirp Analysis control panel.

Enters the Waterfall,Directivity&3D control panel.

Enters the Wavelet Analysis control panel.

Enters the Acoustical Parameters control panel.

Enters the FFT&RTA Analysis control panel.

Enters the Sinusoidal Analysis control panel.

Enters the Multimeter control panel.

Enters the Thiele&Small Parameters control panel.

Enters the Wow&Flutter control panel.

Enters the Leq control panel.

Enters the Linearity&Distortion control panel.

Enters the Loudness Rating calculator.

Enters the Quality Control Processor.

4.4.2 AUTOSCALE

Enables autoscale. When autoscale is active the software, during measurements, determines the optimum Y-scale settings.

4.4.3 HELP

Invokes the Help control panel.

Invokes the Internet On-Line Help.

4.5 HARDWARE CONTROLS TOOLBAR

4.5.1 INPUT CONTROL

channel A input peak meter

Constantly monitors channel A input signal level vs.full digital input scale.

Controls channel A input polarity.

channel A input sensitivity display & control buttons

Displays the actual input sensitivity (in dBV) of the instrument, i.e. the voltage level beyond which the hardware saturates. It is possible to modify it in 10dB steps by pressing the ▼ (**F9**) and/or ▲ (**F10**) buttons.

channel B input peak meter

Constantly monitors channel B input signal level vs.full digital input scale.

Controls channel B input polarity.

channel B input sensitivity display & control buttons

Displays the actual input sensitivity (in dBV) of the instrument, i.e. the voltage level beyond which the hardware saturates. It is possible to modify it in 10dB steps by pressing the ▼ (**SHIFT+F9**) and/or ▲ (**SHIFT+F10**) buttons.

Links input channels full scale level controls. If this button is pressed the two channel sensitivities are set equal and channel A controls act also on channel B.

Selects the Autorange mode. When in autorange mode the input sensitivity is automatically adjusted by the instrument to achieve the optimum signal to noise ratio.

4.5.2 INPUT/OUTPUT LOOPBACK

The CLIO Box features an internal loopback which is very useful for performing self tests.

Connects channel **A** output to channel **A** input with an internal relay.

Connects channel **B** output to channel **B** input with an internal relay.

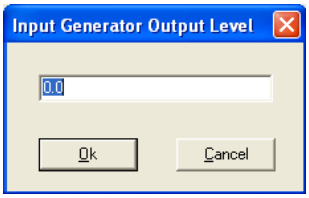
4.5.3 GENERATOR CONTROL

CLIO's generator can be controlled from the dedicated toolbar buttons and dialogs; for a reference about the possible kind of signal you may generate please see chapter 7.

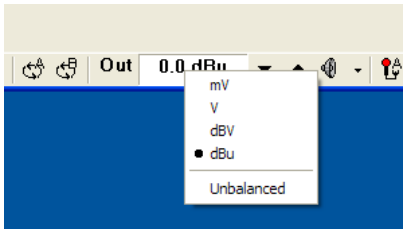
output level display & control buttons

Displays the actual output level (usually in dBu) of the internal generator. This level is valid for both output channels. It is possible to modify it in **1dB step**s pressing the ▼ (**F7**) and ▲ or (**F8**) buttons. If the **Shift** key is pressed simultaneously then the steps are of **0.1dB** increments.

It is also possible to input a numeric value directly with the following dialog which pops up when you **click** on the output level display.



In this case (manual input) the output level will be approximated with a **0.01dB** precision.
If you **right-click** on the output level display you invoke the out units pop up from which it is possible to select the output level unit among dBu, dBV, V and mV.



Checking the Unbalanced option the output level display is referred to the unbalanced outputs of the Clio Box. When this mode is selected the generator output level display is shown in white with black background.

Switches on and off the generator.
Use the **ESC** key to immediately **kill** the generator .
If you wish to receive a confirmation message (Fig.4.3) before playing the generator then check the appropriate box in the **General Options** dialog (5.4).

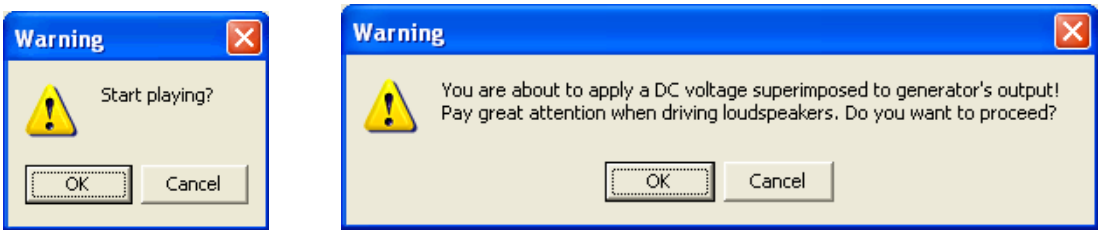


Figure 4.3

generator drop down menu

Clicking on the small arrow beside the generator button will invoke the generator drop down menu, from there it is possible to choose the output signal type to be generated. The default signal at startup is a 1031.25Hz sinusoid.
Refer to Chapter 7 Signal Generator for a detailed description of all generated signals.

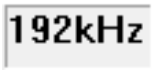
4.5.4 MICROPHONE CONTROL

Switches Channel A 24V phantom power on and off. This supply is capable of operating any balanced microphone and also to operate Audiomatica's microphones MIC-01, MIC-02 and MIC-03 (see later).

Switches Channel B 24V phantom power on and off.

To enter the microphone sensitivity please refer to 5.4 Options.

4.5.5 SAMPLING FREQUENCY



Indicates the current sampling frequency of the instrument. To change it simply click on it and refer to 5.4 Options.

4.6 QCBOX & LPT CONTROLS

Enters the External Hardware Controls dialog box. This dialog box performs controls over some external hardware connected to the computer.

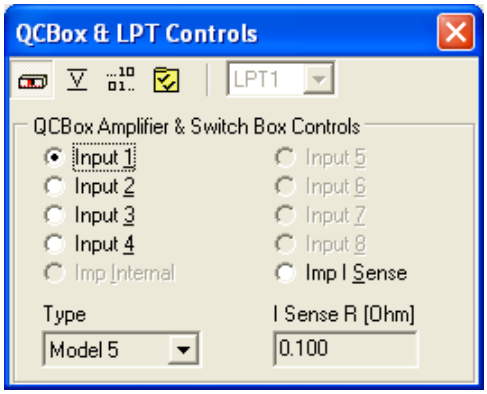


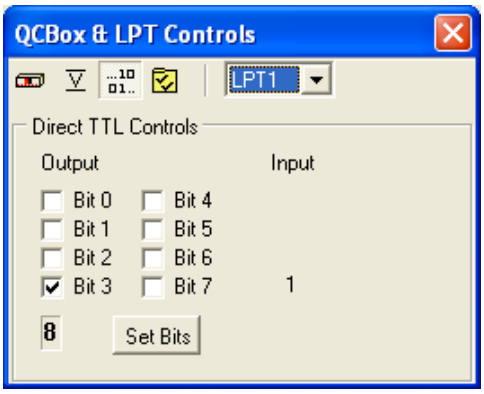
Fig. 4.6 External Hardware Controls dialog box

This control panel helps you when you are operating the CLIOQC Amplifier & SwitchBox.

You may choose the Amplifier & SwitchBox model and set the value of the internal sensing resistor to obtain maximum precision during impedance measurements (for this please refer to chapter 13).

These controls are self-explanatory; they are also covered in the unit's user's manual, in this manual, and everywhere else the amplifier and switchbox is used.

You can also read and write a PC parallel port:



4.6.1 CONTROLLING THE QCBOX 5 POWER AMPLIFIER, SWITCHING AND MEASURING BOX

With this dialog box it is possible to access to the QCBox 5 enhanced features. It is possible to superimpose a DC voltage on the amplifier output, set the microphones phantom voltage and the output current protection threshold. It is also possible to read the Isense DC current and the IN 3 and IN 4 DC voltages.
The digital I/O port is showed on the bottom of the dialog box and monitor the port status, it is possible to write the out bits by simply clicking on them.

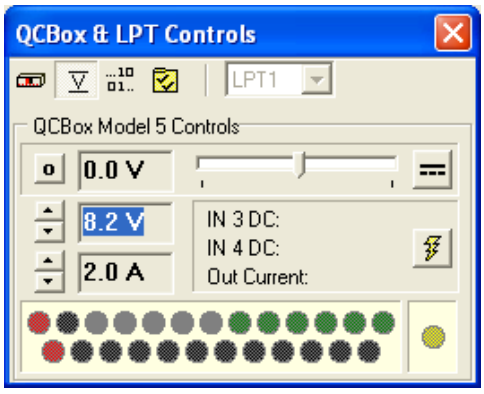


Fig.4.7 QCBOX 5 power amplifier, switching and measuring box control panel

4.7 CONTROLLING TURNTABLES

This control panel allows the control of one or two turntables. **The control of two turntables is available only with the QC software**.
Using two turntables it is possible to measure the loudspeaker response in three dimensions, i.e. the software can send commands to the turntables to aim the loudspeaker under test in a given direction.

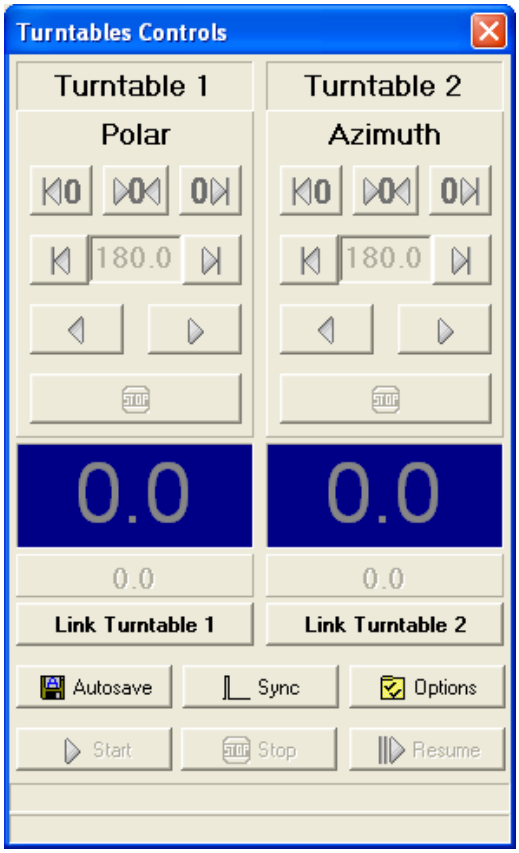


Fig.4.8 Turntables control panel

Reset turntable position to angle 0 by clockwise rotation (degrees up)

Reset turntable position to angle 0 by counterclockwise rotation (degrees down)

Set turntable reference angle (0 degrees)

Goto angle by clockwise rotation (degrees up)

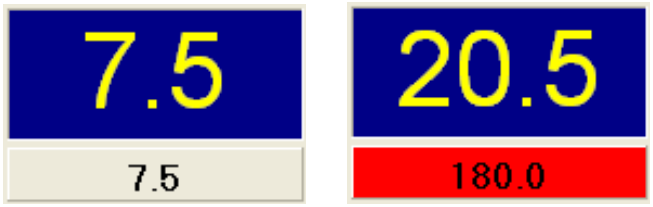
Goto angle by counterclockwise rotation (degrees down)

Step angle by clockwise rotation (degrees up), note that the step size is a turntable setting that cannot be accessed from CLIO

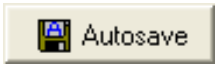
Step angle by counterclockwise rotation (degrees down)

Stop the turntable rotation

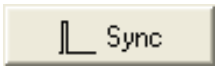
Link Turntable 1 and Link Turntable 2 connect turntables and link the turntable positions to the measurements



Displays turntable current angle (top) and next angle (bottom), while the turntable is rotating the bottom background is highlighted in red.



Open the Autosave Settings dialog



Reset turntable angles according to Autosave Settings

Open the Turntables Option dialog

Start an MLS Autosave measurement set

Halt an MLS Autosave measurement set

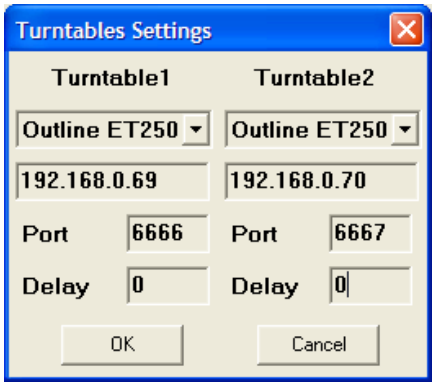
Resume an MLS Autosave measurement set

4.7.1 TURNTABLES OPTIONS DIALOG

With this dialog it is possible to choose which model of turntable to use for each rotating axis (polar and azimuth). The software can take full control of the Outline ET250-3D and the LinearX LT360 turntables. It support also (limited to the polar rotation) a TTL pulse control using the PC parallel port which can be used to trigger the Outline ET/ST turntable or any other device. Using the combo box it is possible to choose which turntable model to use for the polar and azimuth angles (for a definition of polar and azimuth angles please refer to chapter 12).

Outline ET250-3D

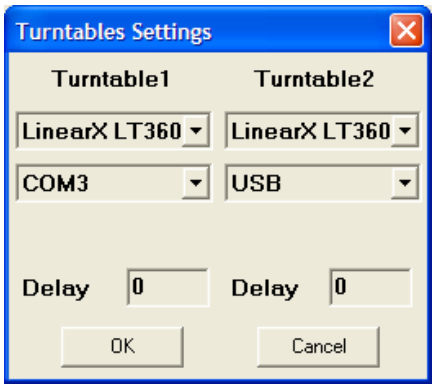
The Outline ET250-3D uses an Ethernet connection, please refer to the manufacturer documentation to setup the device. In the option dialog it is necessary to input the **turntable IP** and **TCP/IP port**.



Note: In order to work properly the basert.dll file must be present into the CLIO installation directory.

LinearX LT360

The LinearX LT360 turntable uses an USB or COM connection, please refer to the manufacturer documentation to the setup of the device. In the option dialog it is needed to input the **communication port** to be used.



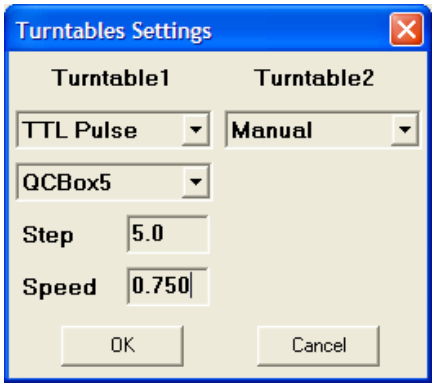
Some turntables settings, such as the rotation speed and the velocity profile must be managed using the software supplied with the turntable. **For correct operations with CLIO software the "Display Readout Polarity" setting of the LT360 turntable must be set on "Unipolar"**.

Note: In order to work properly the lt360lib.dll file must be present into the CLIO installation directory.

The **delay** parameter (in milliseconds) put the software in a wait state after the completion of the turntable rotation, this can be useful in a non anechoic space to let the energy in the room to decay between measurements.

TTL pulse control

Selecting TTL pulse it is possible to control a turntable using a TTL signal from the PC parallel port. **This is valid only for the polar angle and with this selection it is not possible to use two computer controlled turntables.** In this case the second turntable can be only selected as "Manual".



The TTL pulse control uses the parallel port or the output port of a QCBox model V. The information given here apply to the control of the **Outline ET/ST Turntable**; they can be adapted to any other device.

The control is achieved with **Bit 7** of the **parallel port output** bits, as shown in Fig.4.6. The turntable should be connected to the parallel port of the computer by means of a cable defined as follows:

```
PC side DB25 male                ET/ST side DB9 male
Pin 9  <----------------------> Pin 2
Pin 22 <----------------------> Pin 4
All other pins unconnected
```

With the **QCBox5** selected the control is achieved with **Bit 5** of the QC Box output port, the turntable should be connected to the QC Box V output port by means of a cable defined as follows:

```
QCBox5 side DB25 male                 ET/ST side DB9 male
Pin 6  <----------------------> Pin 2
Pin 14 <----------------------> Pin 4
All other pins unconnected
```

The cable should be connected as in the following figure.

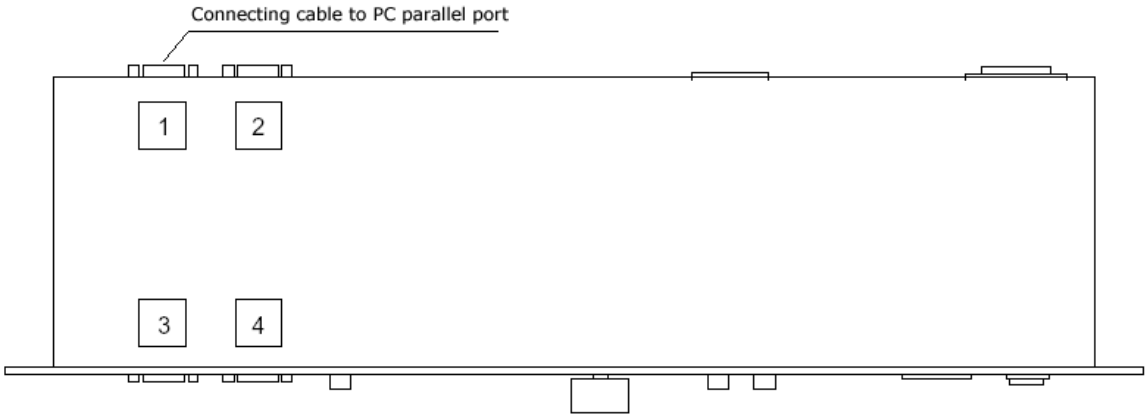


Fig.4.9 Outline ET/ST Turntable connections

Inside the Turntable Option panel it is possible to set its **Resolution** (in degrees) and **Speed** (in rotations per minute); the combination of these settings give the software an indication about how much time to wait after the controlling pulse is output.

Manual

For the azimuth angle only is it possible to choose the "Manual" turntable. This means that the azimuth rotation of the loudspeaker under test must be managed manually. Instead of automatically control the turntable, the software display a pop-up message to the operator and wait for the completion of the manual rotation.

Please refer to chapter 12 for further information on polar measurement sets.

4.8 MAIN MENU AND SHORTCUTS

The user should carefully read this section which gives you a comprehensive list of all the menu and shortcuts available within CLIO. Shortcuts, i.e. keystrokes that you can press to obtain a certain action, will save you time and increase your productivity.

Also refer to Chapter 6 to learn the measurement interface and its associated shortcuts. Note that the measurement specific (MLS submenu, FFT submenu etc.) menus are active only when the measurement control panel is open and selected.

4.8.1 FILE MENU

Please refer to Chapter 5 for a detailed explanation of the File Menu.

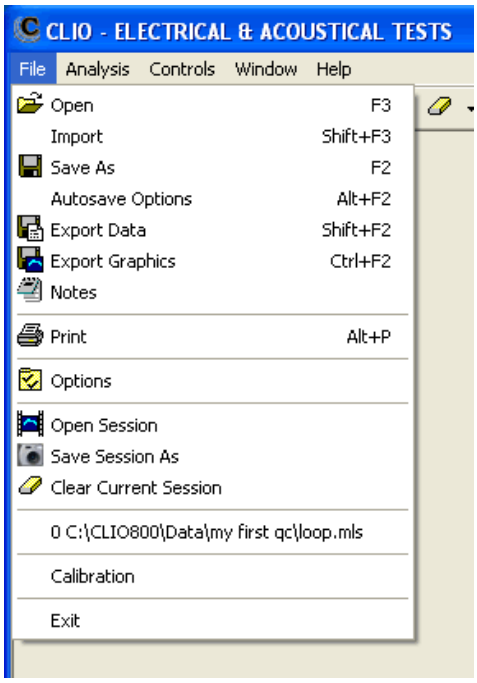


Fig.4.10 File Menu

F3 Saves a measurement file relative to the active control panel.

F2 Loads a measurement file relative to the active control panel.

ALT+F2
Enters the Autosave Settings dialog box (see 5.3.1).

SHIFT+F2
Exports the active measurement as an ASCII file.

CTRL+F2
Exports the active measurement as a Windows Enhanced Metafile, BMP, PNG, JPEG or GIF.

ALT+P
Prints the active measurement.

F6 Enables/disables the AutoScale function (see 4.4.2).

4.8.2 ANALYSIS MENU

The Analysis menu gives you a powerful way to access the measurement menu and menu functions, through the keyboard. Here we present you with all the available menus and shortcuts; the shortcuts, when present, are visible from each submenu and are listed on the right of the function; refer to the specific chapters for each measurement for details about it.

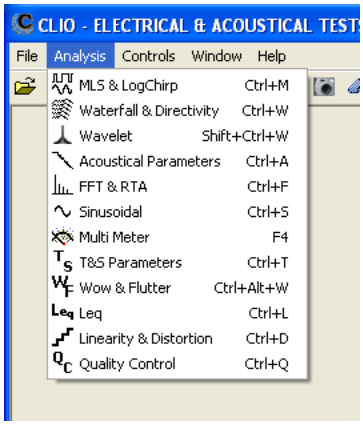


Fig.4.11 Analysis Menu

CTRL+M
Enters the MLS&LogChirp Analysis control panel.

CTRL+W
Enters the Waterfall&Directivity control panel.

SHIFT+CTRL+W
Enters the Wavelet Analysis control panel.

CTRL+A
Enters the Acoustical Parameters control panel.

CTRL+F
Enters the FFT&RTA Analysis control panel.

CTRL+S
Enters the Sinusoidal Analysis control panel.

F4
Enters the Multimeter control panel.

CTRL+T
Enters the Thiele&Small Parameters control panel.

CTRL+Alt+W
Enters the Wow&Flutter control panel.

CTRL+L
Enters the Leq control panel.

CTRL+D
Enters the Linearity&Distortion control panel.

CTRL+Q
Enters the Quality Control Processor.

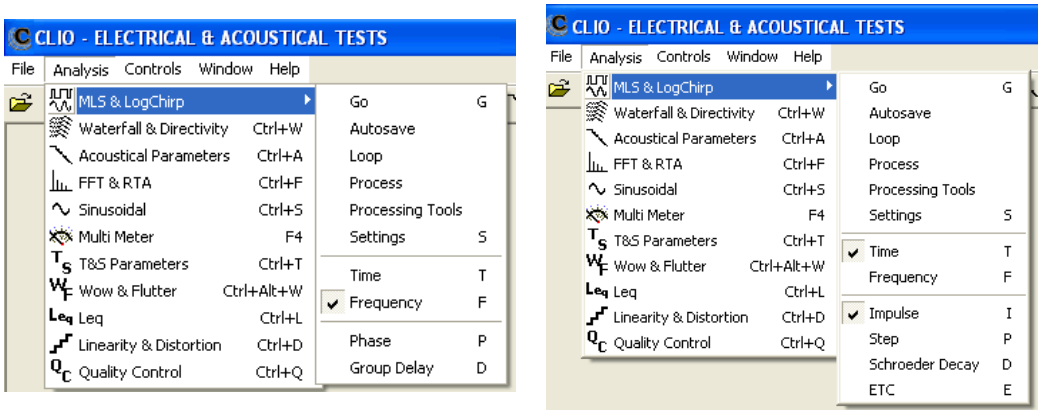


Fig.4.12 MLS&LogChirp Submenu - Frequency and Time

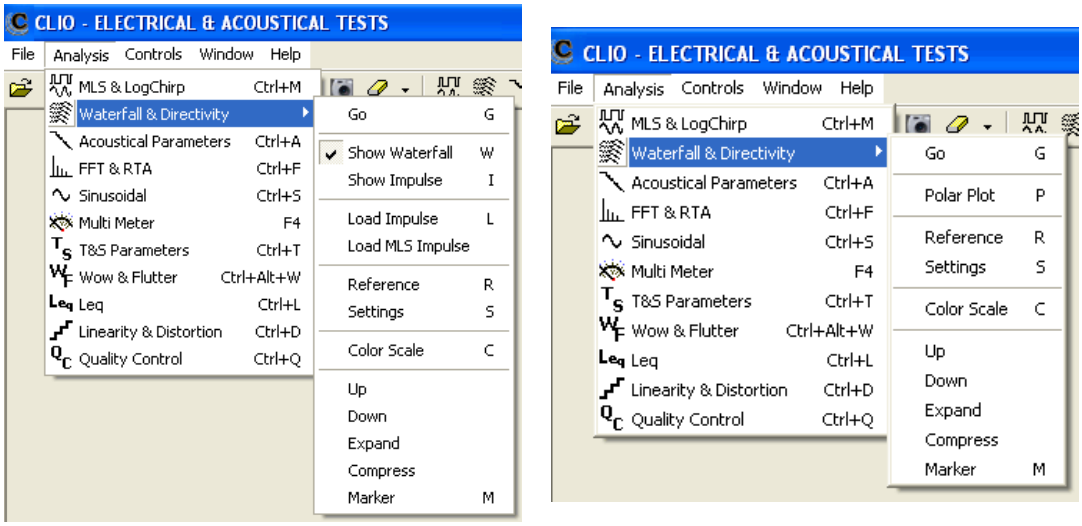


Fig.4.13 Waterfall and Directivity Submenu

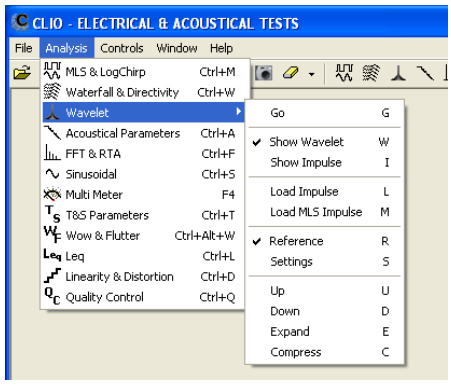


Fig.4.14 Wavelet Analysis SubMenu

Fig.4.15 Acoustical Parameters SubMenu

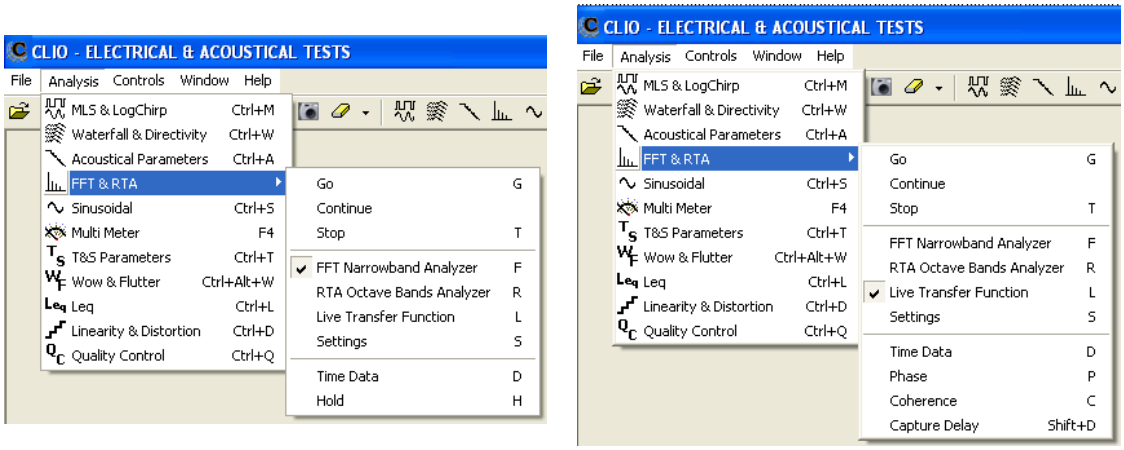


Fig.4.16 FFT and FFT Live SubMenu

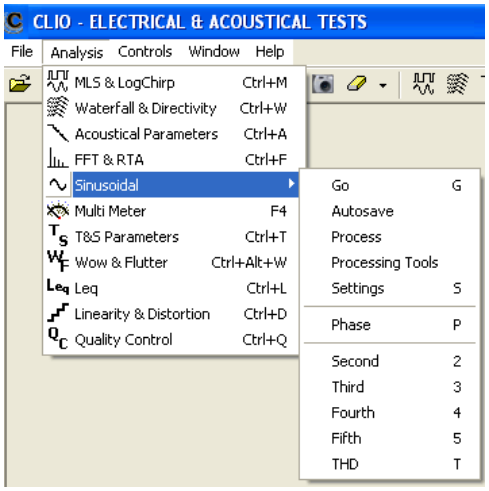


Fig. 4.17 Sinusoidal Submenu

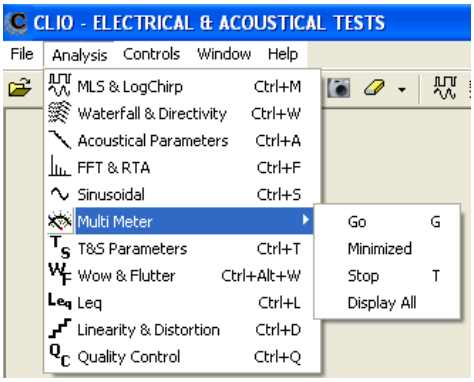


Fig. 4.18 Multi-meter Submenu

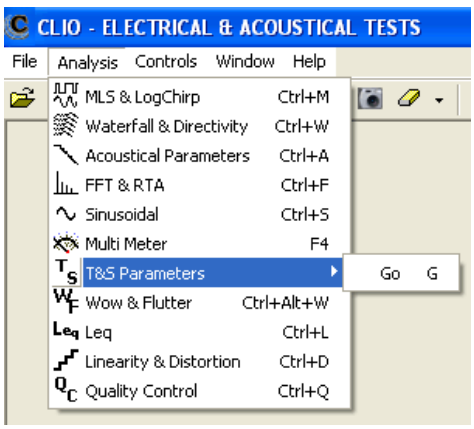


Fig. 4.19 T&S Parameters Submenu

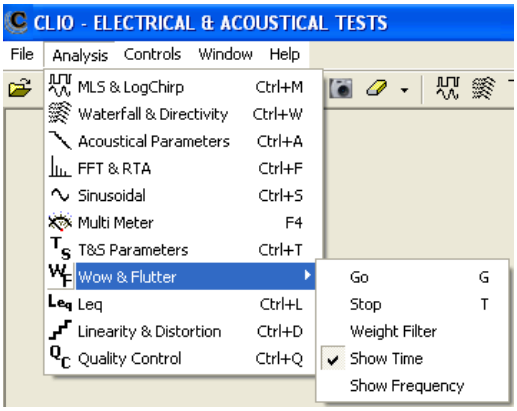


Fig. 4.20 Wow&Flutter Submenu

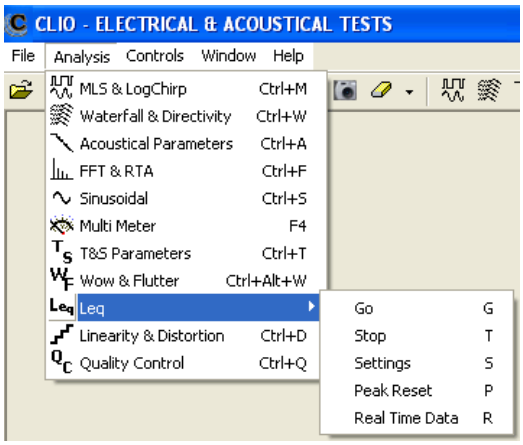


Fig. 4.21 Leq Submenu

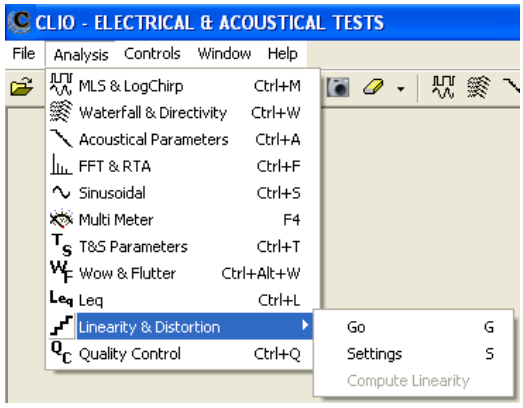


Fig. 4.22 Linearity&Distortion Submenu

4.8.3 CONTROLS MENU

The Controls Menu is the heart of your CLIO hardware; learn how to access hardware control through the keyboard in detail. Refer also to 4.4.2, 4.4.3, 4.4.4 and 4.5.

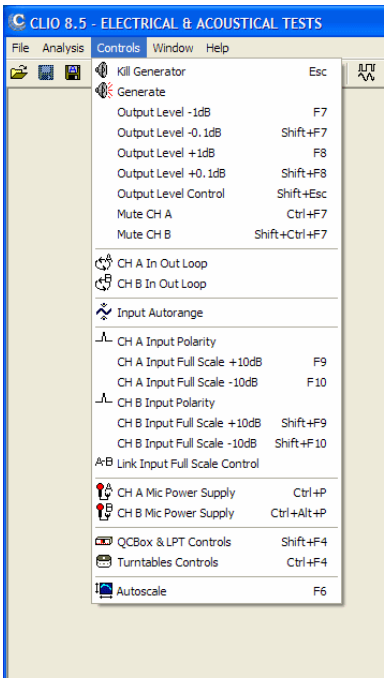

CLIO 8.5 - ELECTRICAL & ACOUSTICAL TESTS
File
Analysis
Controls
Window
Help
Kill Generator Esc
Generate
Output Level -1dB F7
Output Level -0.1dB Shift+F7
Output Level +1dB F8
Output Level +0.1dB Shift+F8
Output Level Control Shift+Esc
Mute CH A Ctrl+F7
Mute CH B Shift+Ctrl+F7
CH A In Out Loop
CH B In Out Loop
Input Autorange
CH A Input Polarity
CH A Input Full Scale +10dB F9
CH A Input Full Scale -10dB F10
CH B Input Polarity
CH B Input Full Scale +10dB Shift+F9
CH B Input Full Scale -10dB Shift+F10
Link Input Full Scale Control
CH A Mic Power Supply Ctrl+P
CH B Mic Power Supply Ctrl+Alt+P
QCBox & LPT Controls Shift+F4
Turntables Controls Ctrl+F4
Autoscale F6

Fig. 4.23 Controls Menu

ESC

Immediately kills the generator. Equivalent to releasing

F7 Decreases the output level of 1dB. Equivalent to ▼

SHIFT+F7

Decreases the output level of 0.1dB. Equivalent to SHIFT+ ▼

F8 Increases the output level of 1dB. Equivalent to ▲

SHIFT+F8

Increases the output level of 0.1dB. Equivalent to SHIFT+ ▲

F10

Increases channel A input acceptance of 10dB. If the Link Button is pressed then increases also channel B input acceptance of 10dB. Equivalent to ▲

SHIFT+F10

Increases channel B input acceptance of 10dB. Equivalent to ▲

F9 Decreases channel A input acceptance of 10dB. If the Link Button A-B is pressed then decreases also channel B input acceptance of 10dB. Equivalent to ▼

SHIFT+F9

Decreases channel B input acceptance of 10dB. Equivalent to ▼

CTRL+P

Swithces channel A microphone power supply on and off. Equivalent to

CTRL+ALT+P

Swithces channel B microphone power supply on and off. Equivalent to

SHIFT+F4

Enters the QCBox and LPT control panel. Equivalent to

CTRL+F4

Enters the Turntables Controls panel. Equivalent to

F6 Enables autoscale. Equivalent to

4.8.4 WINDOWS MENU

The Windows Menu helps you manage all opened windows (i.e. measurement control panels) in a standardized way. You can Tile or Cascade the open windows or access each one directly.

Fig. 4.24 Windows Menu

4.8.5 HELP MENU

From the Help Menu you can access all the available help resources installed in your computer or available directly from Audiomatica over the internet.

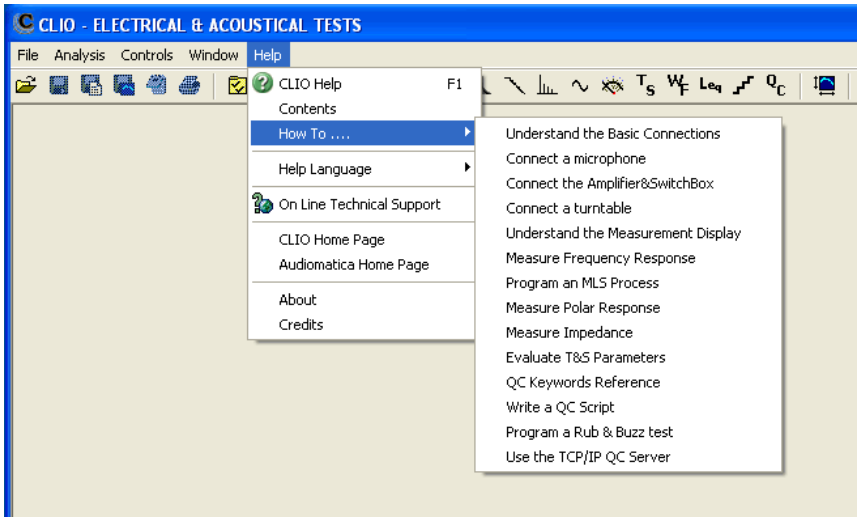


Fig. 4.25 Help Menu

F1 Invokes the On-Line Help.

4.9 BASIC CONNECTIONS

In order to correctly interface CLIO with the outside world you should always keep in mind the following electrical specifications:

MAXIMUM INPUT VOLTAGE:	+40 dBV (283 V peak-to-peak)
MAXIMUM OUTPUT VOLTAGE:	+18dBu (6.156Vrms) (sine)
INPUT IMPEDANCE:	128 kOhm
OUTPUT IMPEDANCE:	660 Ohm

4.9.1 CONNECTING THE CLIO BOX

The CLIO system is stereo and can simultaneously process two balanced analog I/O channels which are named channel **A** and **B**. The output of channel **B** is usually driven in parallel with channel **A** output.

The CLIO Box input uses two XLR Combo female connectors. These particular connectors accept also a 6.3mm standard jack.

The CLIO Box output uses two XLR male connectors. In parallel you also find two RCA plugs that are used to take the output signal unbalanced.

Please note that there it will always be a 6 dB difference between the CLIO balanced and unbalanced outputs.

By default the **Output Level display** is referred to the **balanced output**, in this case the **unbalanced output level is 6 dB lower**. In order to display the unbalanced output level is necessary to select the option in the Output Level display; in this case the balanced output is 6dB higher than the unbalanced one.

Figure 4.26

On the rear panel you also find a digital **SPDIF** output.

Unless you are carrying out impedance measurements with the **Internal Mode** selected, one of CLIO outputs will usually be connected to an external power amplifier that will drive the loudspeaker or to the electronic apparatus or other system under test.
The output of the system under test will be connected to one of the CLIO inputs.

4.9.2 CONNECTING A MICROPHONE

For acoustical measurements, the microphone (optionally followed by a preamplifier or power supply) requires to be connected to CLIO's input channel.

When using a MIC-01, MIC-02 or MIC-03 microphone it is possible to connect it **directly** to CLIO's input; remember, in this case, to switch the phantom voltage on by pressing the phantom button . It is good practice to wait a few seconds before taking measurements as the microphone's output stabilizes.

If the measuring point is far from the PC, always lengthen the connection between the preamplifier and CLIO. Make sure that you never use microphone cable that is longer than the one that has been supplied.

In Fig. 4.28 we see the typical test setup for performing acoustical measurements of a loudspeaker. Please **note** that in this schematic diagram the output of the power amplifier is connected to the loudspeaker with an inversion in the cables polarity; this compensates the fact that **microphones are usually phase inverting** ; when making polarity measurements always treat the measuring chain in this respect considering that the CLIO hardware itself is NON-INVERTING and that all calibrations are usually made under this assumption: any external device like amplifiers, microphones, accelerometers, preamplifiers etc. has to be carefully checked.

It is possible to achieve the same result, via software, simply clicking on the **input polarity button** in the hardware controls toolbar.

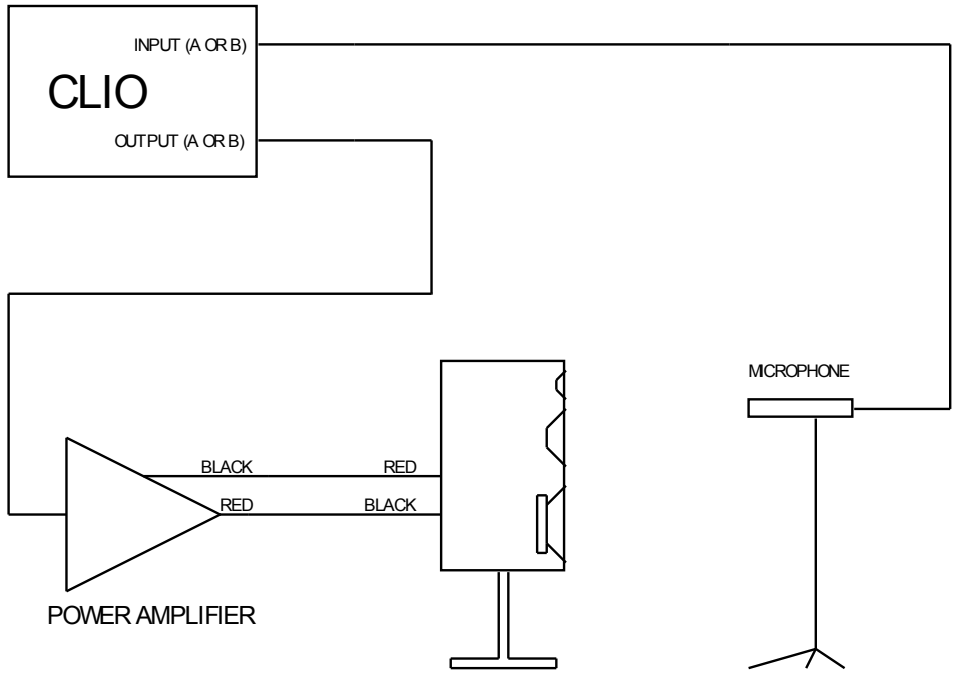


Figure 4.28

4.9.3 CONNECTING THE CLIOQC AMPLIFIER & SWITCHBOX

Fig.4.29 and Fig. 4.30 show the connections of a CLIOQC Amplifier & SwitchBox to CLIO.

The unit has its internal switcher set for response measurements.

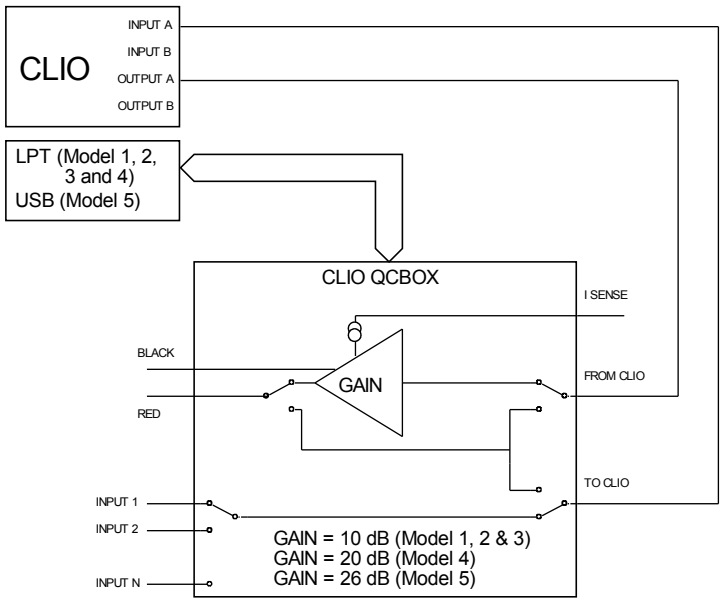


Figure 4.29

To take impedance measurements use the ISense mode (see chapter 13) as the ImpInternal mode is not available when connecting to the FW-01 balanced input.

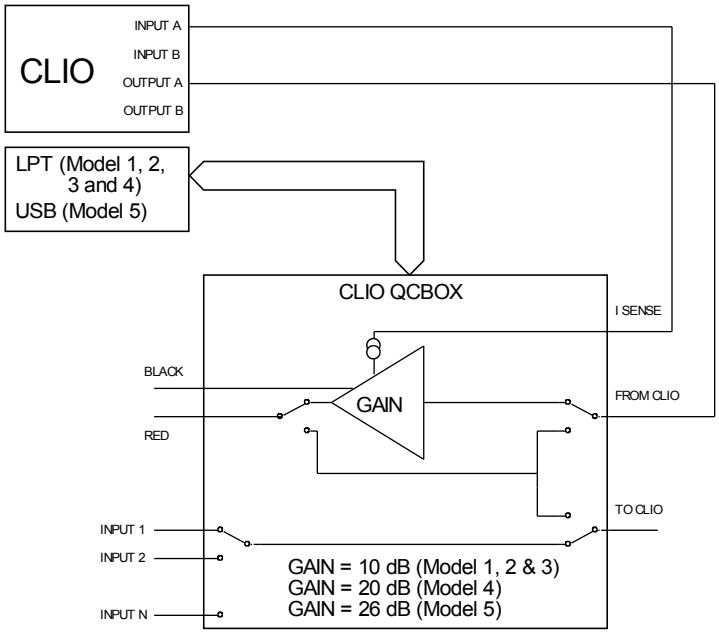


Figure 4.30

5 SYSTEM OPERATIONS AND SETTINGS

5.1 INTRODUCTION

This chapter completes the introduction to CLIO started in Chapter 4.
Here you will find information about:

- **Files extensions**
- **File operations**
- **Exporting data**
- **Exporting graphics**
- **Printing**
- **Software option**
- **Desktop control**
- **Calibration of CLIO**
- **Startup options**
- **Measurements settings**

5.2 REGISTERED FILE EXTENSIONS

During its installation CLIO registers several file extensions which will let you easily find a file done during your work. Browsing your hard disk you will then encounter the icons that we are going to describe.

MLS&LogChirp data files.

MLS&LogChirp process files.

Waterfall,Directivity & 3D data files.

Wavelet data files.

Acoustical Parameters data files.

FFT and RTA data files.

Sinusoidal data files.

Sinusoidal process files.

Multi-meter data files.

T&S parameters data files.

Wow&Flutter data files.

Leq data files.

Linearity&Distortion data files.

Multitone definitions files.

Autosave definitions files.

Desktop snapshot files.

CLIO setup files.

OLD CLIO Signal files; not used but supported.

OLD MLS&LogChirp impedance data files; not used but supported.

OLD Sinusoidal impedance data files; not used but supported.

When you find a CLIO data file it is possible to invoke the program simply clicking on the file itself; if CLIO is open it will load the file in the appropriate menu and display it, if it is closed it will be launched then the file opened.
If you click on a **Desktop snapshot file** you will recall a saved work session comprising open menu and data (see 5.5).

It is possible to run only a single instance of CLIO.

5.3 FILE MENU AND MAIN TOOLBAR BUTTONS

Fig. 5.1 shows the File menu and the Export submenu. Refer to 4.5.1 for the shortcuts active.

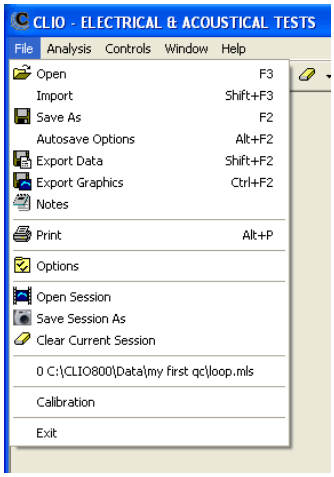


Fig.5.1 File Menu

5.3.1 LOADING AND SAVING FILES

Loads a measurement file relative to the active control panel. It is important to note that it is possible to load more than one data file type from the following menu:
- **MLS** loads frequency response files (*.mls) and impedance response files (*.mlsi)
- **FFT** loads FFT files (*.fft) and CLIO4 RTA files (*.rta)
- **Sinusoidal** loads frequency response files (*.sin), impedance response files (*.sini), CLIO4 sinusoidal frequency response files (*.frs) and CLIO4 impedance response files (*.imp).
You can select the desired file type from the 'Files of type' drop down inside the Open dialog box.

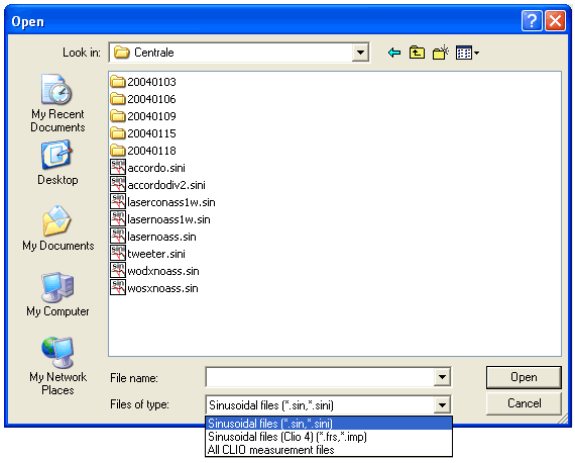


Fig.5.2 Open dialog

Saves a measurement file relative to the active control panel. It is important to note that the following menu saves more than one data file type:
- **MLS** saves frequency response files (*.mls), impedance response files (*.mlsi) or impulse response as wave files (*.wav)
- **Sinusoida**l saves frequency response files (*.sin) and impedance response files (*.sini)
- **Leq** saves Leq analysis files (*.leq) and captured data wave files (*.wav)

Invokes the Autosave Settings dialog. It is possible to define the autosaving rules that will be followed by the measurements capable of this feature (MLS and Sinusoidal). There are two different autosave modes: 1D and 3D.
1D mode:

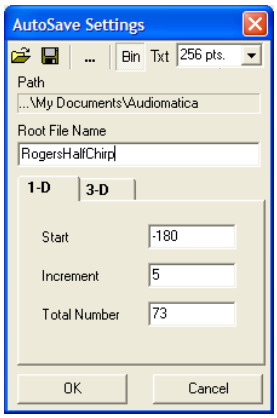


Fig.5.3a Autosave settings - 1D mode

There are five settings which serve to define the autosaved file name:
Path defines the folder where the file will be saved; it is possible to choose it clicking on the browse for folder button (...). In Fig.5.3a we see path defined as My Documents\Audiomatica
Root File Name defines the part of the file name that will not change during autosave; in Fig.5.3a it is 'RogersHalfChirp'.
Start defines the initial number appended to the root. This number will always be multiplied by 100 to give space for two decimals management. Numbers are appended with spaces. In Fig.5.3 start is -180; this will define the first autosaved file name as 'My Documents\Audiomatica\RogersHalfChirp -18000'
Increment defines the increment to be given to the autosaved file names. In the example of Fig.5.3 the second autosaved file will be named 'My Documents\Audiomatica\RogersHalfChirp -17500'
Total Number defines the number of autosaved files after which the process is automatically ended
It is possible to choose to save in the standard binary file format (Bin) and/or to export in text format (Txt) (see also 5.3.2); the drop down chooses the number of export data points for MLS files.
It is possible to save (and load) these definitions in particular files called Autosave Definition Files (*.asd).

3D mode:
There are five settings which serve to define the autosaved file name:
Path defines the folder where the file will be saved; it is possible to choose it clicking on the browse for folder button (...). In Fig.5.3b we see path defined as My Documents\Audiomatica
Root File Name defines the part of the file name that will not change during autosave; in Fig.5.3b it is 'RogersHalfChirp'.
Polar Start defines the polar angle start
Polar Step defines the polar angle step
Polar Stop defines the polar angle stop
Azimuth Start defines the azimuth angle start
Azimuth Step defines the azimuth angle step
Azimuth Stop defines the azimuth angle stop
Please refer to 12.8.1 to a the definition of polar and azimuth angles.
It is possible to choose to save in the standard binary file format (Bin) and/or to export in text format (Txt) (see also 5.3.2); the drop down chooses the number of export data points for MLS files.
It is possible to save (and load) these definitions in particular files called Autosave Definition Files (*.asd).

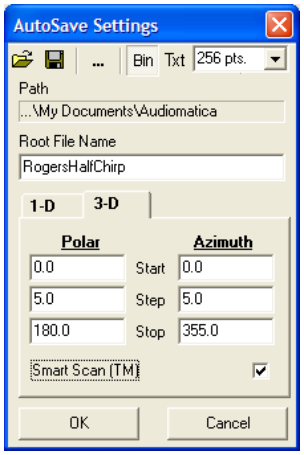


Fig.5.3b Autosave settings - 3D mode

5.3.2 EXPORTING DATA

CLIO is able to export the currently active measurement in an ASCII file (*.txt).

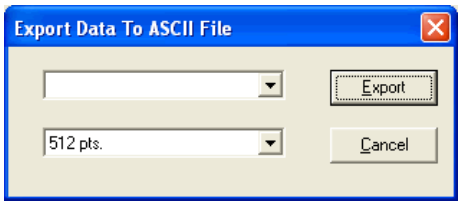


Fig.5.4 Export dialog

Upon performing this choice you will be prompted by the Export dialog. Depending on the measurement menu you are working with, it will be possible to choose different data to export. Here is a list of the possibilities, menu by menu:

MLS > Display Frequency Data
The program will export the frequency data as they are displayed on the screen, with resolution variable from 256 to 2048 points, regardless the current MLS size.

MLS > FFT Frequency Data
The program will export the frequency data with the resolution of the current MLS size.

MLS > Time Data
The program will export the acquired impulse response or active post process.

FFT > Display Frequency Data
The program will export the processed frequency data as they are displayed on the screen, with its internal resolution of 2048 points, regardless the current FFT size.

FFT > FFT Frequency Data
The program will export the processed frequency data with the resolution of the current FFT size.

FFT > Last FFT Data
The program will export the last calculated FFT frequency data with the resolution of the current FFT size.

FFT > Last Time Data
The program will export the last captured time data with the resolution of the current FFT size.

Sinusoidal > Frequency Data
The program will export the acquired frequency data with their fraction of octave frequency resolution.

Sinusoidal > Frequency + Distortion Data
As above plus THD and R&B data.

T&S > T&S Parameter

Linearity&Distortion > Distortion data

Wow&Flutter > Wow&Flutter data

5.3.3 EXPORTING GRAPHICS

CLIO is able to create **enhanced metafiles** (*.emf), **bitmaps** (*.bmp), **portable network graphics** (*.png), **JPEG** (*.jpg) or **GIF** (*.gif) of the currently active measurement. The graph is drawn using the same color of printouts; you can define them with the Options dialog, see 5.3.6.

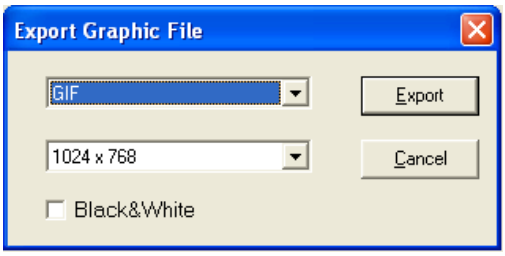


Fig.5.5 Export Graphics dialog

Check the **Black&White** box to discard color information.

5.3.4 PRINTING

Enters the Notes dialog where it is possible to input comments to be saved with the actual measurement and inspect other measurement information. Check boxes enable printing notes and exporting notes to graphics files.

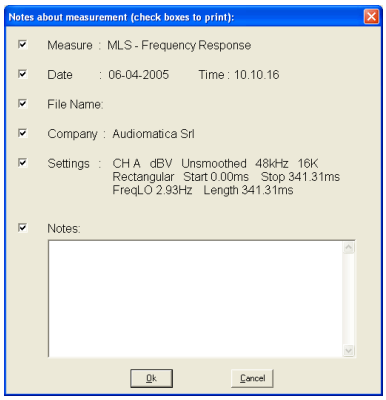

Notes about measurement (check boxes to print):
Measure : MLS - Frequency Response
Date : 06-04-2005 Time : 10.10.16
File Name:
Company : Audiomatica Srl
Settings : CH A dBV Unsmoothed 48kHz 16K
Rectangular Start 0.00ms Stop 341.31ms
FreqLO 2.93Hz Length 341.31ms
Notes:
Ok
Cancel

Fig.5.6 Notes dialog

Prints the current active measurement. The definition of printing colors is done with the Setup dialog, see 5.3.5.

5.4 OPTIONS

Opens the CLIO Options dialog box (Fig.5.7)

5.4.1 GENERAL

Opening this tab (Fig. 5.7) you can define the following:
- The **Company Name** which will appear in all printouts.
- Some **On Exit** settings regarding when the program has to prompt and if you want to autosave and reload the measurement session (see 5.5).
- The **Signal Generator prompts**.
- The location of the **Hardware Controls Toolbar**.
- Some behavior of the graphic curve display (see Chapter 6).

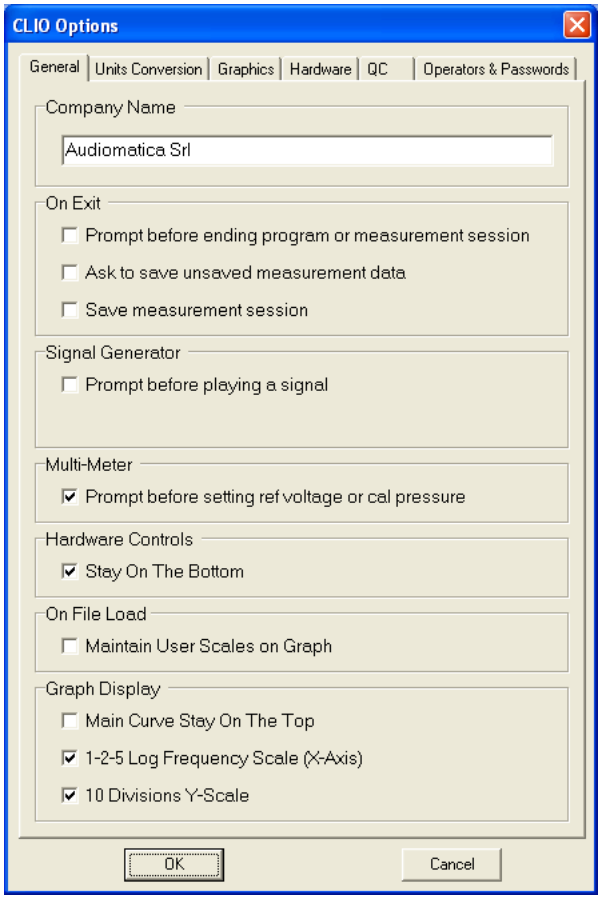


Fig.5.7 Options General dialog

5.4.2 UNITS CONVERSION

Opening this tab you can define the following:
- Enter the microphone sensitivity and the microphone response correction.
- Enter all other transducers sensitivity and reference levels.

MICROPHONE SENSITIVITY

When taking acoustical measurements, the readings and the scales will be in Pascals (Pa, dBSPL, dBPa or dBPa/V). In this case, the software will assume that you are measuring a pressure quantity and it therefore needs to know a conversion factor that defines the voltage produced by the microphone when it is measuring a certain pressure. This conversion factor is usually the sensitivity of the microphone (as found in the microphone's calibration chart) or the sensitivity of the microphone + preamplifier chain of equipment.

If you are working with the CLIO system standard accessories there are two possible cases:

a) you may use a microphone MIC-01, MIC-02 or MIC-03 directly connected to FW-01; it is necessary to input the sensitivity of the microphone (in mV/Pa) **increased of 3.3dB i.e. multiplited by 1.465**.

b) you are using the PRE-01 preamplifier, it is necessary to know its internal gain; if it is 0dB then input the microphone sensitivity, if it is +20dB then input the microphone sensitivity multiplied by 10.

NOTE: It is necessary to input two separate sensitivities, one for channel **A** and one for channel **B**.

Please also refer to 8.4.1 for a description of the procedure for assessing the sensitivity of the microphonic chain using an acoustic calibrator capable of producing 94dBSPL.

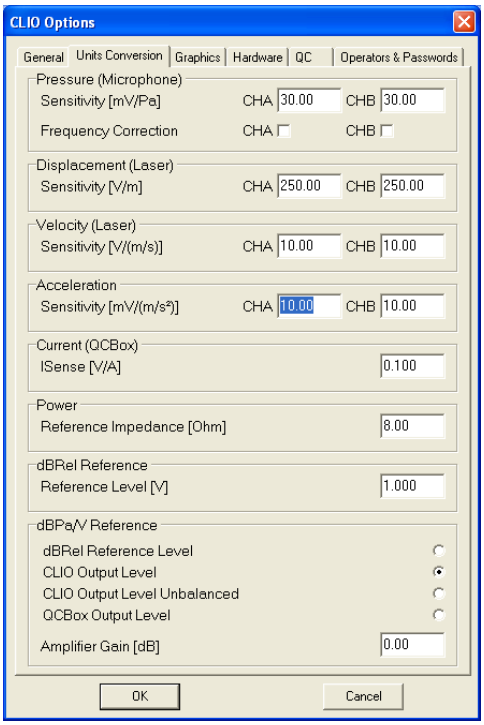


Fig.5.8 Options Units Conversion

CORRECTING THE MICROPHONE RESPONSE

By checking the Microphone Correction check boxes the software will correct the measured curve according to the data stored in two text files named "MICA.CAL" (for input channel A) and "MICB.CAL" (for input channel B).

The microphone correction files, if present, must be placed inside the installation folder (usually c:\program files\audiomatica\clio 10\).

Note: The maximum number of correcting points allowed is 100.

The example below shows a sample text file created to store the microphone frequency response:

```
Freq     dB     Phase
1000     0        0
4000     0.25     0
8000     0.33     0
10000    0.5      0
15000    1.75     0
20000    2.5      0
```

Note: microphone correction will affect the measured curve only if dBSPL (or dBPa or dBPa/V) is selected as Y Scale unit.

If your microphone came with frequency calibration data supplied by Audiomatica (see 2.2.3) you can find the correction file inside the furnished disk.

The file is stored with the name ***serialnumber.cal*** ; for example in the disk given with the microphone with serial 9501255 you will find the "9501255.CAL" file.

To use it you must:

1) Copy it inside the CLIO 10 installation folder

2) Rename it according to which input channel it is connected; "MICA.CAL" for channel A or "MICB.CAL" for channel B.

DISPLACEMENT, VELOCITY, ACCELERATION, CURRENT, POWER

Set the sensitivity for each of these units where used in the program.

dBREL REFERENCE

Sets the voltage reference for the unit dBRel.

dBPa/V REFERENCE

When measuring with dBPa/V it is possible to use as reference level:
- the dBRel reference voltage
- CLIO output level (balanced)
- CLIO output level (unbalanced)
- QCBox Output
- Output of any amplifier of a given gain in dB connected to the balanced output

5.4.3 GRAPHICS

Opening this tab you can define the following:
- Screen Colors
- Print (and graphics export) colors
- Screen line width
- Print (and graphics export) line width and font size.

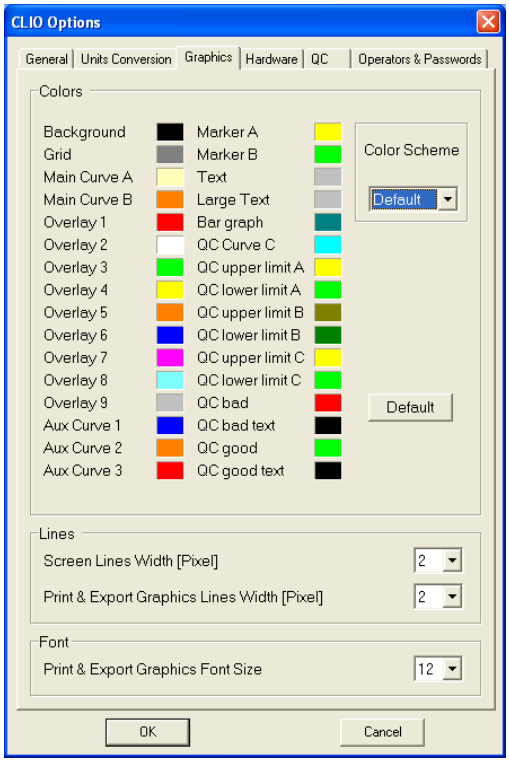


Fig.5.9a Options Graphics dialog

Apart from the Default color scheme, which is not changeable, it is possible to load and customize up to 6 different color schemes: Classic (for CLIO 6 users), User1, User2, User3, User4 and Printing. The **Printing** color scheme, as the name implies, will affect your printouts (and exported graphic files) and, if selected, will let you preview how they appear on your screen.

Default button
When defining a color scheme you may press the Default button which will load the Default scheme for your reference. If you are modifying the printing color scheme it will load the default printing colors.

5.4.4 HARDWARE

Within this tab you can select the **sampling frequency** of the FW-01 unit (see 4.5.6). It is possible to choose either 48kHz, 96kHz or 192kHz.

5.4.5 QC AND OPERATORS AND PASSWORDS

In case of QC software version there are also two other tabs **QC** and **Operators & Passwords**.

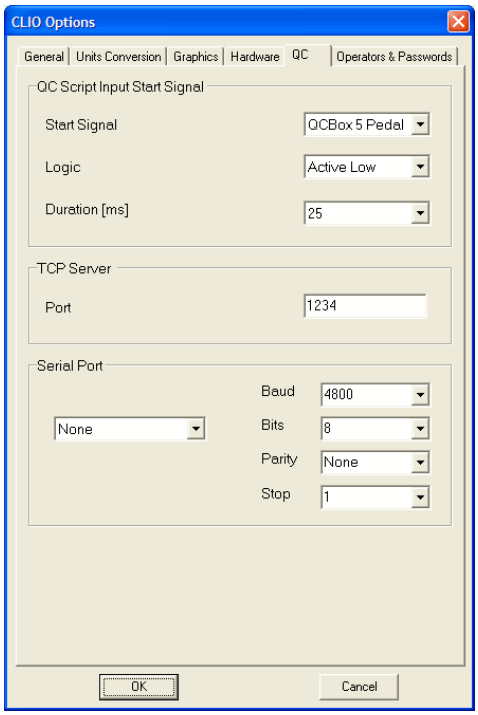


Fig.5.9b Options QC

Opening the QC tab you can define the following:
- select the source and logic of the external QC script start signal
- select the **serial port** used to control external RS-232 devices
- select the **TCP port** to be used by the QC measurement server.

Operation of the QC Operators & Passwords tab is treated in detail in the QC manual (see 19.4.8).

5.5 DESKTOP MANAGEMENT

Desktop management is a powerful feature that lets you save your work at a certain point and reload exactly as it was.

It is possible to do this automatically when exiting CLIO; at successive startup the program will automatically reload from where you left; to do this activate the **Save measurement session** from CLIO General Options.

Load a previously saved measurement session (*.sna files).

Takes a snapshot of current measurement session and saves it to disk (*.sna files).

If pressed clears current measurement desktop i.e. closes and resets all measurement menu. It is also possible to clear one single measurement selectively opening and choosing from the associated drop down menu.

5.6 CALIBRATION

File > Calibration

This option will perform a calibration of your CLIO hardware. Please refer to paragraph 3.7 and follow the procedure described.

In order to determine, at any given time, if it is necessary to calibrate CLIO do the following:
- Let the system warm up
- Proceed to perform the verification described in 3.7.1
- Consequently decide whether or not to calibrate

The result of the measurement may vary in some way from the time we calibrated because of many small changes in measurement conditions, including changes in the atmospheric conditions, the season, and the mains voltage.

Note: the CLIO hardware is highly precise and stable and, under normal operating conditions, does not require frequent calibrations.
Always perform a calibration if:
- CLIO asks for it showing the "System Not Calibrated" message
- You reinstalled CLIO in a different computer
- You installed a software upgrade

5.7 STARTUP OPTIONS AND GLOBAL SETTINGS

You can start CLIO directly clicking on the **CLIO.exe** executable that is saved in the installation directory (usually C:\Program Files\Audiomatica\CLIO 10); you may also access CLIO either from **Start Menu>Programs>CLIO 10** or creating a shortcut on your Desktop.

A second way to run CLIO is to click on a registered file; in this way you will not only run the program but also load the file into the appropriate measurement menu.

CLIO creates a **temporary folder** for several uses. This folder is located under the Common Application Data folder with the following path (Windows XP):
C:\Documents and Settings\All Users\Application Data\Audiomatica\CLIO 10\Temp
under Windows Vista or 7: C:\ProgramData\Audiomatica\CLIO 10\Temp

CLIO relies, during startup, on a configuration file named **cliofw.stp** which resides in the temporary folder.

This file is written each time the program ends and saves several settings that will be reloaded and reconfigure your system.

Among them we find (see chapters 4 and 5):

- program Options
- generator output level
- input sensitivity
- phantom power supply state
- autorange state
- microphone settings
- CLIOQC Amplifier & SwitchBox
- color scheme and other setup settings
- main window state
- global reference level
- autoscale state

Note: It is possible to return the system to its initial default state (after installation) by deleting the cliofw.stp file.

5.7.1 SAVING MEASUREMENT SETTINGS

Measurements settings can be saved from within the various measurement menu; to do this simply check the Save Settings box in the Settings dialog of each menu (Fig.5.10); refer to specific menu chapters for details on settings.

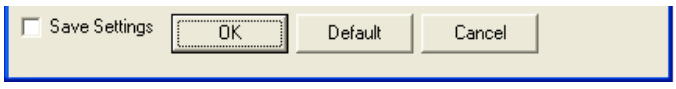


Fig.5.10 Save Settings check box

Settings are saved in the **mls.stp** (MLS&LogChirp), **sin.stp** (Sinusoidal), **fft.stp** (FFT), **wtf.stp** (Waterfall&Directivity), **acp.stp** (Acoustical Parameters) , **lin.stp** (Linearity&Distortion) , **leq.stp** (Leq) files inside the temporary folder.

Upon finding one of these files CLIO will reset the corresponding menu to the saved settings.

6 COMMON MEASUREMENT INTERFACE

6.1 INTRODUCTION

This chapter deals with the graphical user interface which is used to display and manage the measured curves within all CLIO frequency measurement menus. In particular this Common Measurement Interface (CMI) is used by the FFT, MLS and Sinusoidal menu. The understanding of CMI behavior and capabilities is very important to use CLIO at its best.

6.2 UNDERSTANDING THE DISPLAY IN FRONT OF YOU

Fig.6.1 explains the main objects found in a frequency response measurement display.

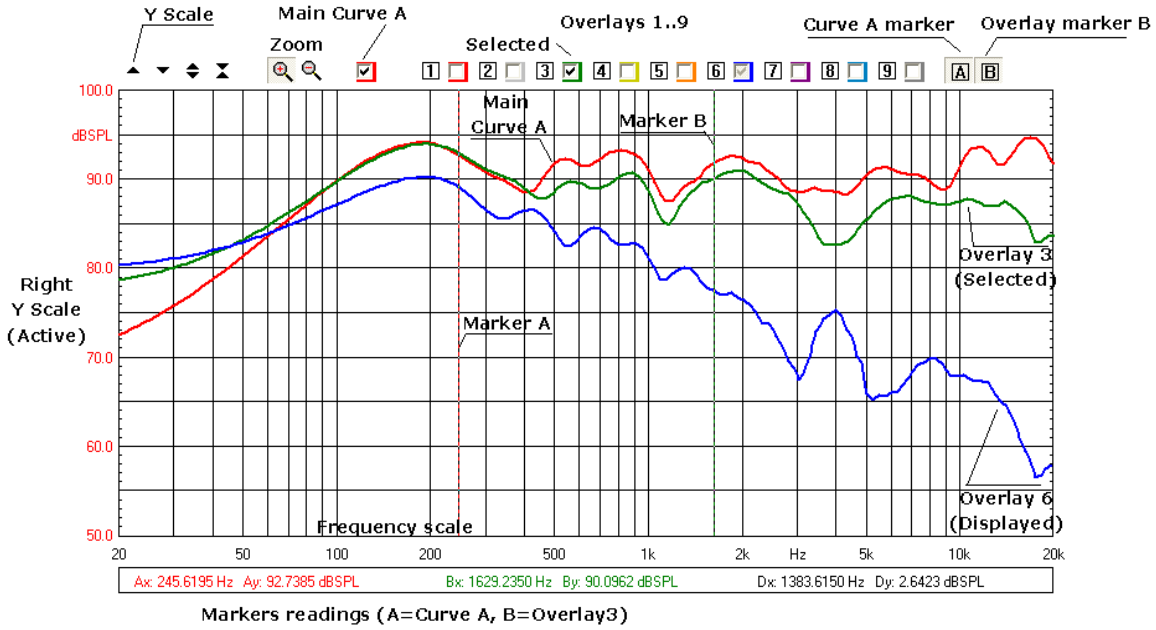


Figure 6.1

Inside the graph you find the **main curve A** which reflects an executed (or loaded from disk) measurement; up to nine **overlays curves** which are stored by the user and can be controlled interactively, the two **markers** which are activated by clicking on the respective buttons.

Above the graph itself we find several buttons and checkboxes which divide into three main categories: Y scales controls, main curve, zoom and overlays management.

Each overlay can be displayed, hidden or selected with the relative checkbox.

Marker A reads curve A; marker B has a twofold operation: it reads the selected overlay, if present, otherwise reads again curve A.

Aside and below the graph we find the Y Scales, the Frequency (or Time) Scale and Marker Indicators. The Y scale is of the same color of the main curve selected.

The objects described may, from case to case, not all be present at the same time,

as in the case of Time Data display in the FFT menu. The frequency (or time) scale may be logarithmic or linear. A particular representation is the MLS time domain which will be discussed later in 6.6.
It is possible to have two graphs in the same control panel (see FFT). In this case one is referred as active after you have clicked on it.

To change the colors of the screen, main curve and overlays refer to section 5.4.

6.2.1 STEREO MEASUREMENTS DISPLAY

Fig.6.2 shows the differences that are present when a stereo measurement is taken or loaded from disk.

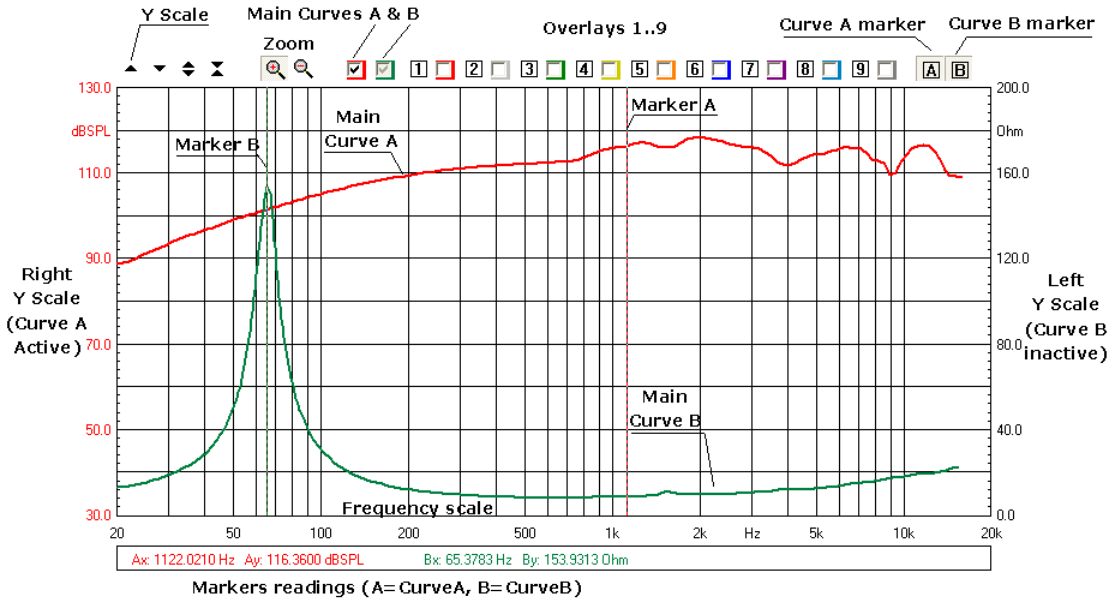
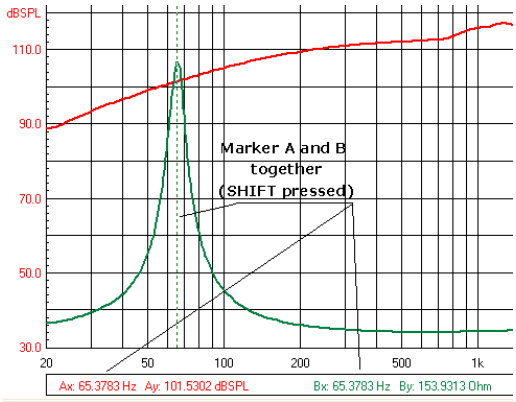


Figure 6.2

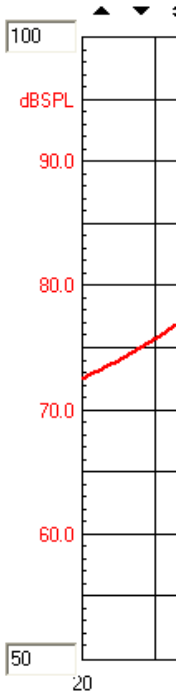
Now you find two **main curves A and B** with two appropriate checkboxes to control them; you may display, hide or select each of the main curves. By selecting a main curve the relative Y scale is activated and scale controls operate on it.
If no overlay is present and selected marker B reads the main curve B, otherwise it reads the selected overlay.

6.2.2 COLLAPSING MARKERS

If you hold the SHIFT key pressed while moving the markers with the mouse you will obtain that the two markers collapse into a single one reading the same frequency point.



6.2.3 DIRECT Y SCALES INPUT



It is possible to direct input of the Y scales values; to activate the input boxes simply click on the scale extremes.

6.3 BUTTONS AND CHECKBOXES

Moves (shifts) the selected curve upward.

Moves (shifts) the selected curve downward.

Expands (magnifies) the selected curve; it also changes the Y scale respectively.

Compresses (reduces) the selected curve; it also changes the Y scale respectively.

Zooms the curve in; it is possible to execute multiple zoom in actions.

Zooms out the curve completely i.e. returns to the default initial zoom state.

Switches the main curve A on and off. In case of stereo measurements displays, hides or selects the main curve A.

If present, for stereo measurements, displays, hides or selects the main curve B

Stores the main curve selected into overlay 1. It also displays overlay 1.

Stores the main curve selected into overlay 2. It also displays overlay 2.

Stores the main curve selected into overlay 3. It also displays overlay 3.

Stores the main curve selected into overlay 4. It also displays overlay 4.

Stores the main curve selected into overlay 5. It also displays overlay 5.

Stores the main curve selected into overlay 6. It also displays overlay 6.

Stores the main curve selected into overlay 7. It also displays overlay 7.

Stores the main curve selected into overlay 8. It also displays overlay 8.

Stores the main curve selected into overlay 9. It also displays overlay 9.

Displays, hides or selects the respective overlay. It also shows its color.

Enables marker A.

Enables marker B.

6.4 HOW TO ZOOM

1) Click on the Zoom+ button.

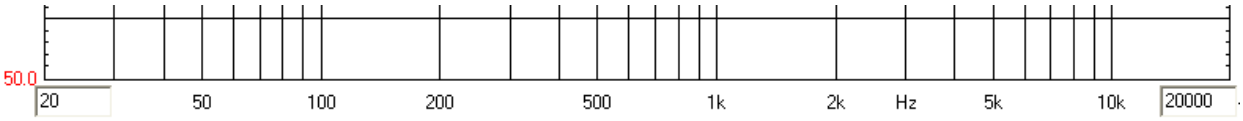
2) Position the mouse and PRESS the left mouse button at the beginning of your selection and **keep the mouse button pressed**. Don't just click otherwise you get a warning message!

3) With the mouse button pressed move the mouse until the second selection point.

4) Only now release the left mouse button.

Be careful: you must have the button pressed from point 2) to point 4)!

It is possible to zoom by direct input of the frequency extremes of the scale; to activate the input boxes simply click on the scale extremes.



6.5 SHORTCUTS AND MOUSE ACTIONS

The following keystrokes and mouse actions are active:

up arrow
equivalent to on the active graph

Shift+up arrow
equivalent to on the active graph

down arrow
equivalent to on the active graph

Shift+down arrow
equivalent to on the active graph

mouse click
activates the graph; useful when more than one graph is present (see FFT)

mouse left down
activates the marker

mouse left drag
moves the marker

mouse wheel up
equivalent to

mouse wheel down
equivalent to

6.6 THE MLS TIME DOMAIN DISPLAY

When entering the MLS&LOG CHIRP (but also Waterfall or Acoustical Parameters) time domain you will find a different display (Fig. 6.2).

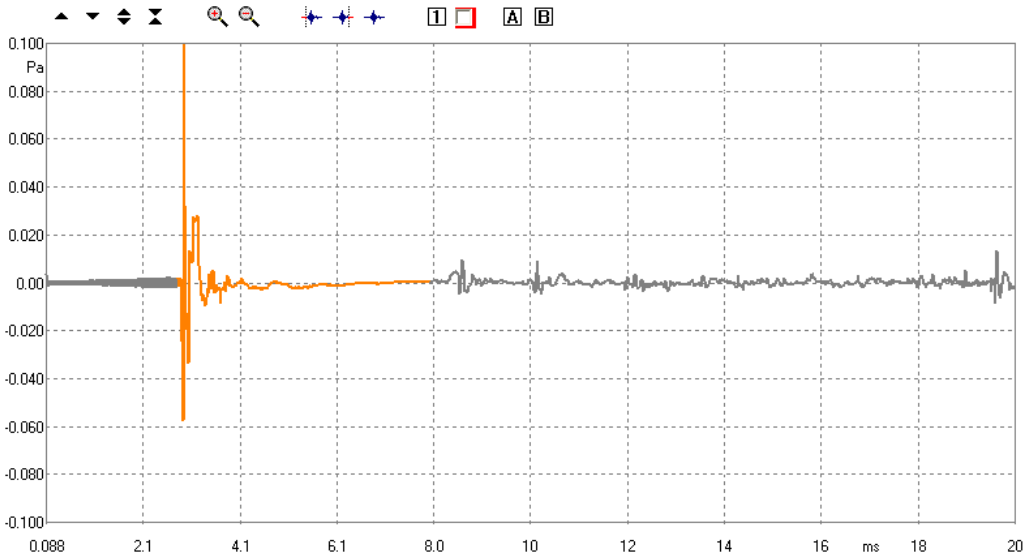


Figure 6.2

In this case there is only one overlay. It is also possible to **select** a portion of the main curve by means of three particular buttons. The selected portion of the main curve is identified by a **start** and **stop** point and is drawn in a different color from the unselected portion.

Defines the **start point** of the selection. **Before clicking with the button activate Marker A and position cursor to the desired point.**

Defines the **stop point** of the selection. **Before clicking with the button activate Marker A and position cursor to the desired point.**

Returns the curve to a completely unselected state.

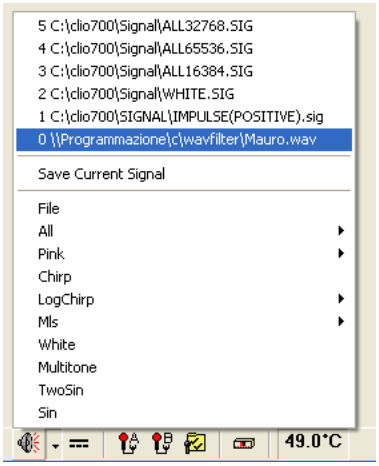
7 SIGNAL GENERATOR

7.1 INTRODUCTION

This chapter deals with the programmable signal generator of CLIO. Each paragraph explains a type of signal, its settings and gives a time frequency analysis obtained with the FFT narrowband analyzer (see chapter 9).

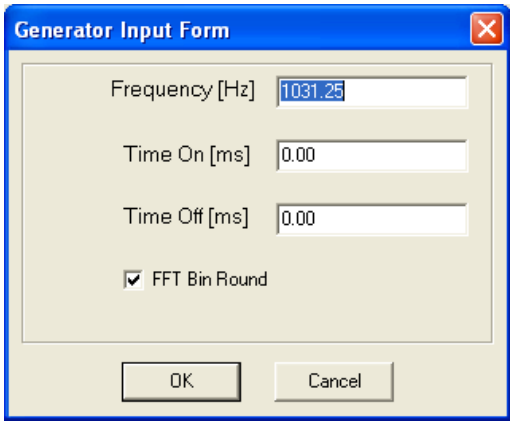
Refer also to 4.5.3 for all hardware and software controls associated with the signal generator.

Clicking on the generator button drop down you access the signal generator menu.



7.2 SINUSOID

It is possible to generate sinusoids of given frequency. Select the Sin choice in the generator menu.

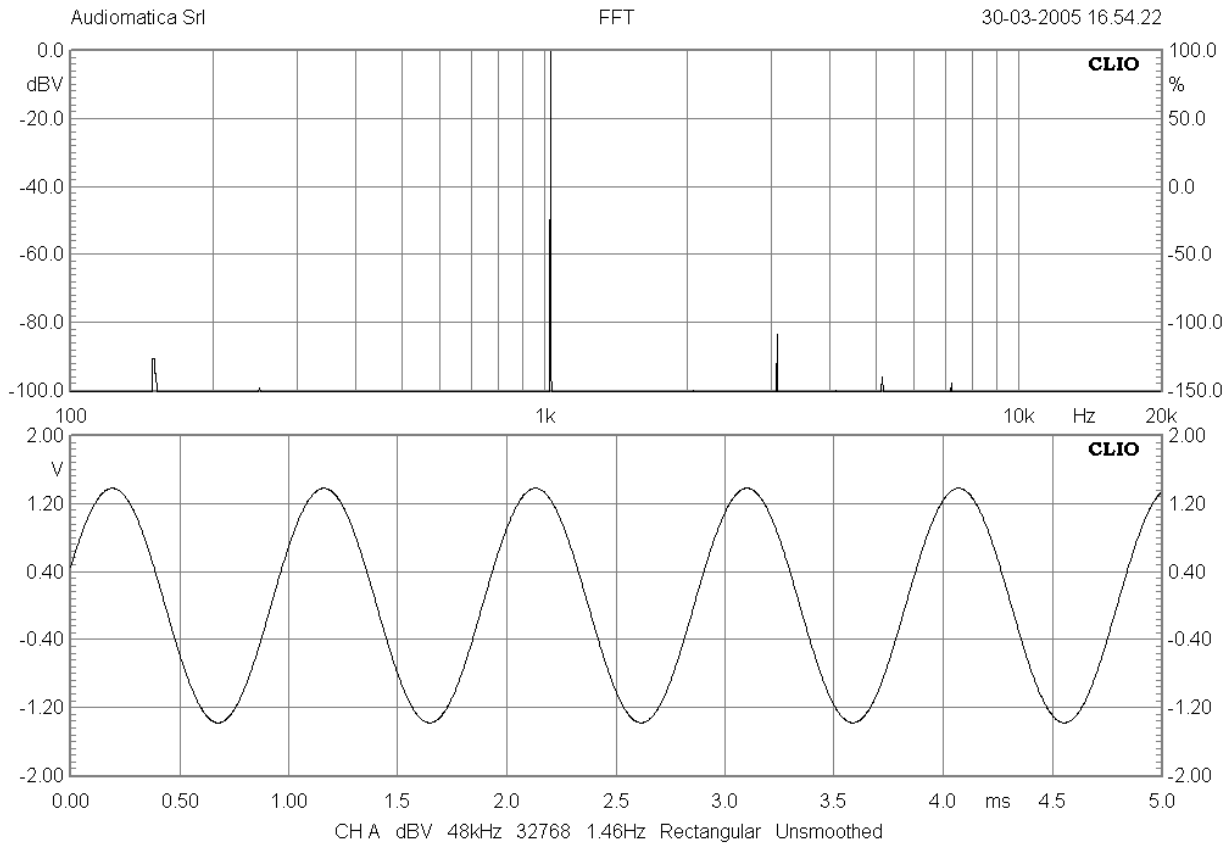


The sinusoid can be **continuous**; leave the two inputs Time On and Time Off at zero.

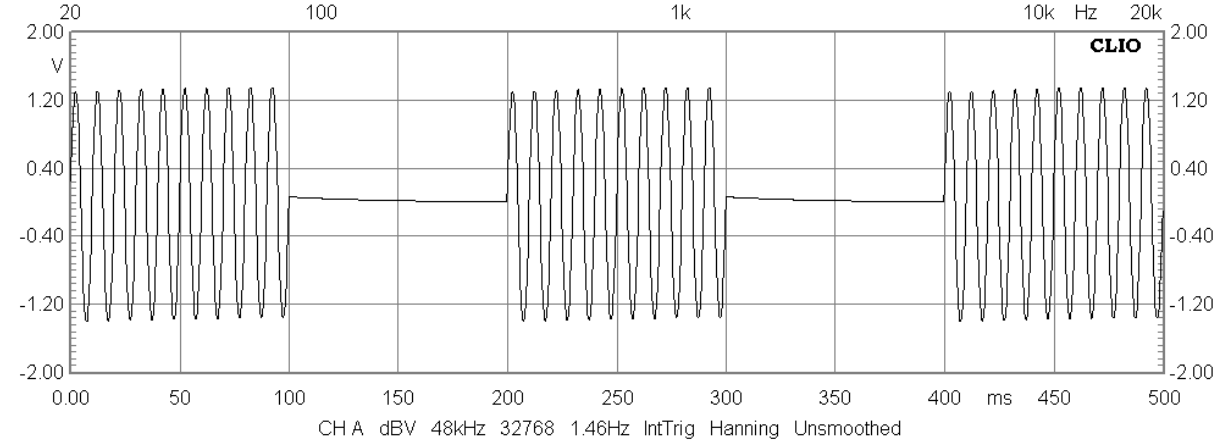
Or it can be **bursted**; input the desired values in the Time On and Time Off inputs.

Select **FFT Bin Round** if you desire that the frequency is approximated to the nearest FFT bin (with respect to the actual FFT size setting). Please refer to 9.5 for a detailed explanation of this feature.

The following figure shows a 1031.25Hz continuous sinusoid.

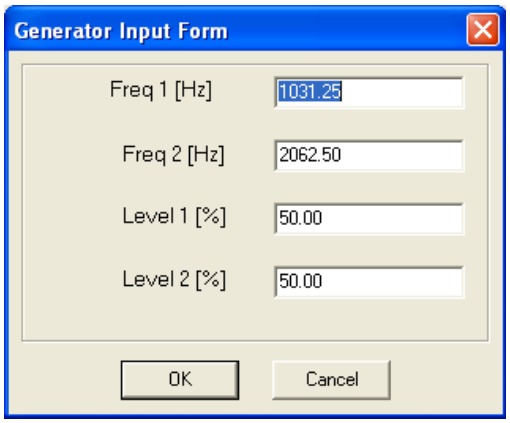


The following figure shows a 100Hz bursted sinusoid.

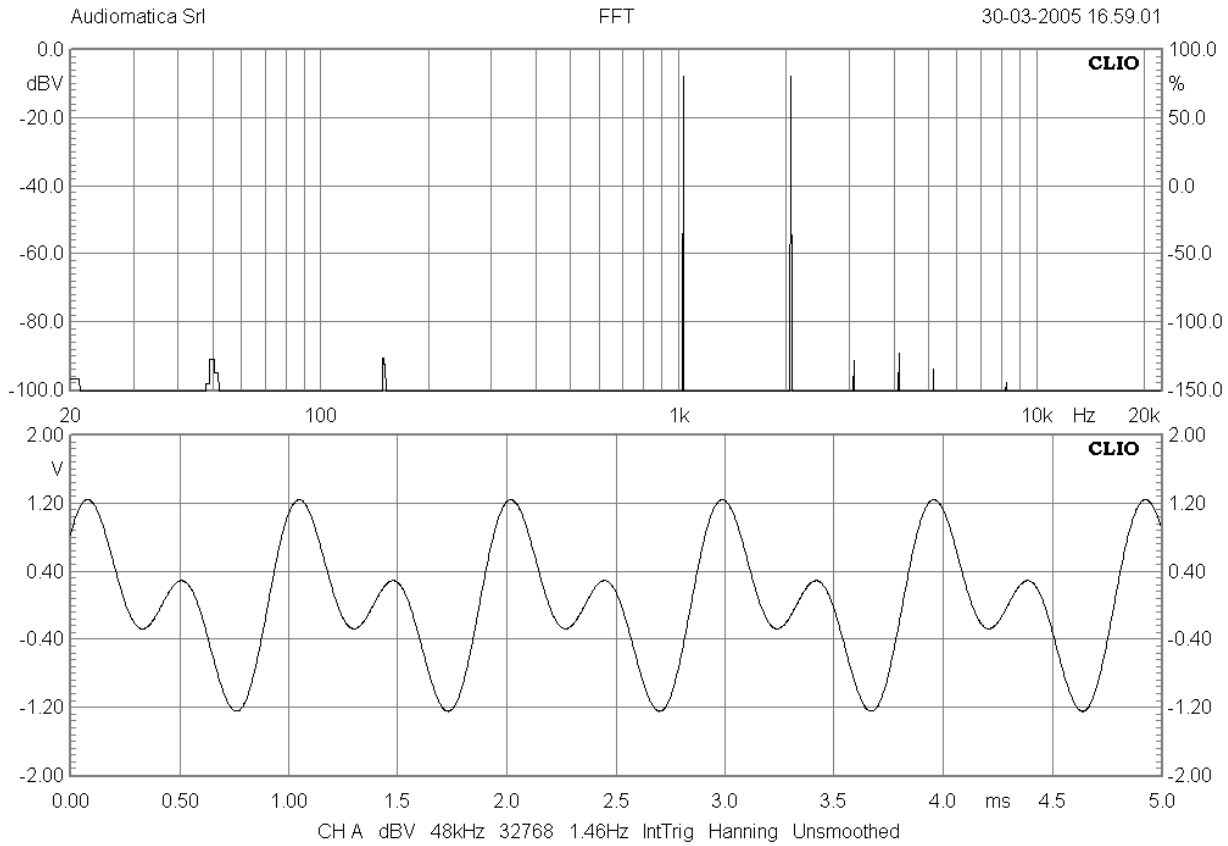


7.3 TWO SINUSOIDS

It is possible to generate two sinusoids of given frequencies and amplitudes. Select the TwoSin choice in the generator menu.

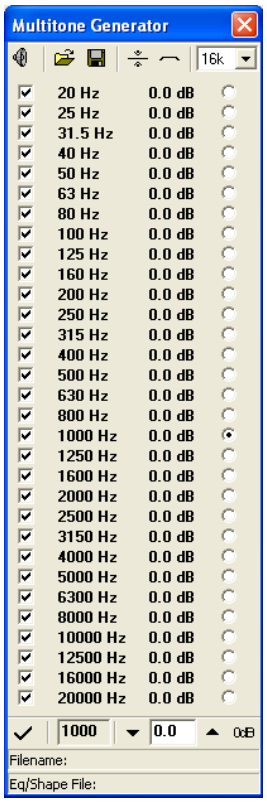


The following figure shows a signal consisting of a 1031.25Hz and 2062.5Hz of same amplitude (50% each).

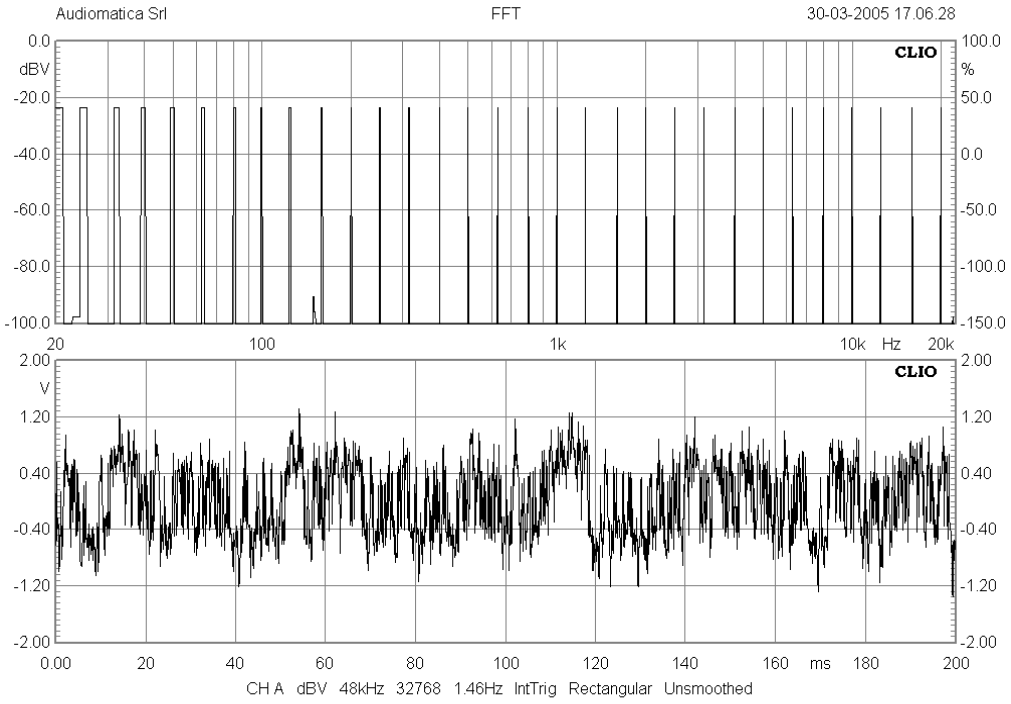


7.4 MULTITONES

It is possible to generate multitones (mutiple sinusoids signals). Select the Multitone choice in the generator menu.



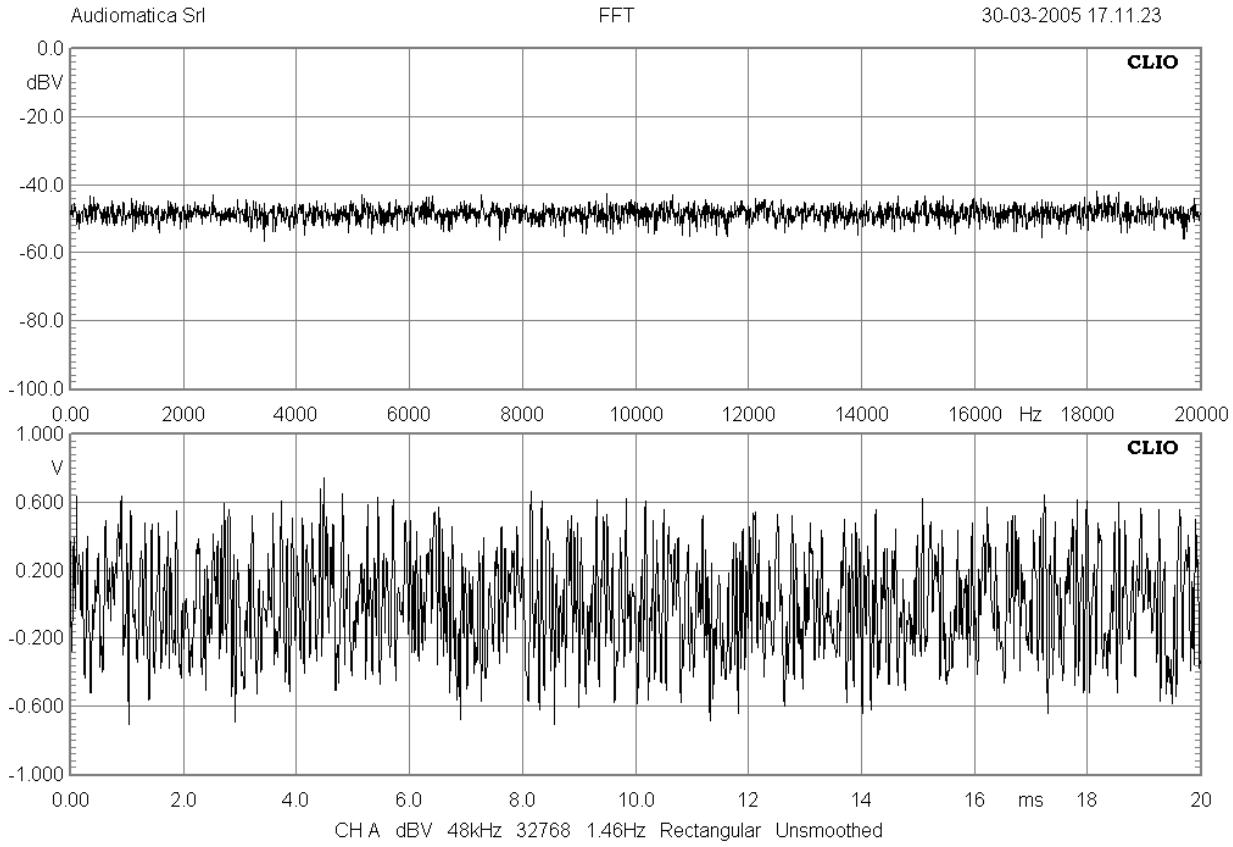
The following figure shows a multitone signal consisting of 31 sinusoids each with frequency corresponding to the center frequencies of the standard 1/3rd of octave bands from 20Hz to 20kHz and same amplitude.



7.5 WHITE NOISE

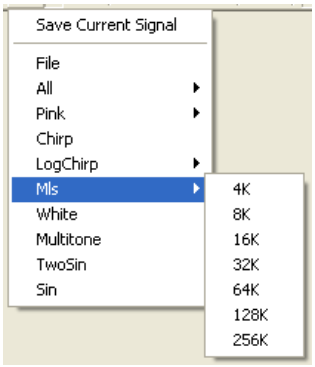
It is possible to generate a white noise. Select the White choice in the generator menu.

The following figure shows the white noise signal.



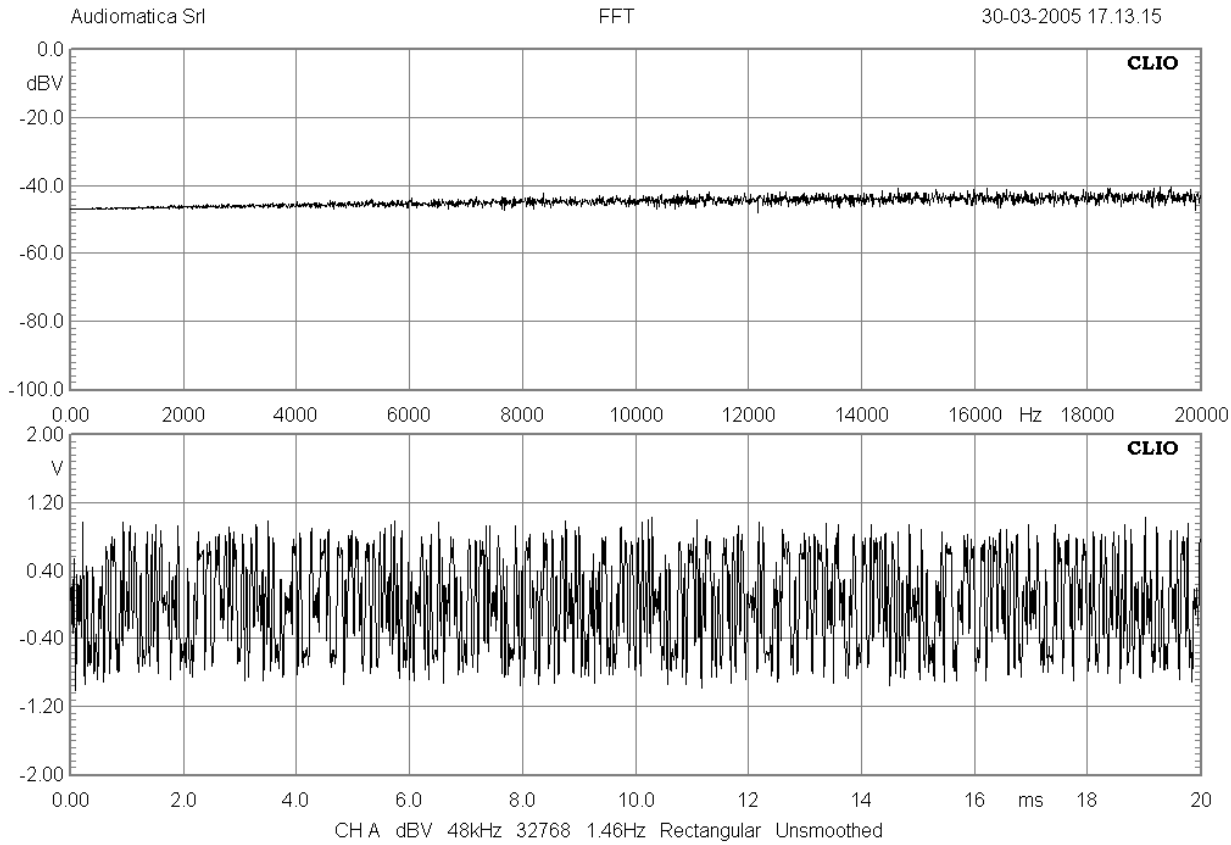
7.6 MLS

It is possible to generate MLS (maximum length sequences) of given length. Select the MLS choice in the generator menu.



These signals are the same used in the MLS analysis menu and should be used to test them.

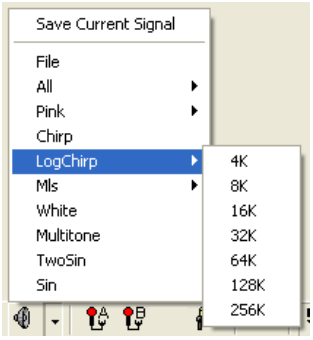
The following figure shows a MLS signal of 32k length.



7.7 CHIRPS

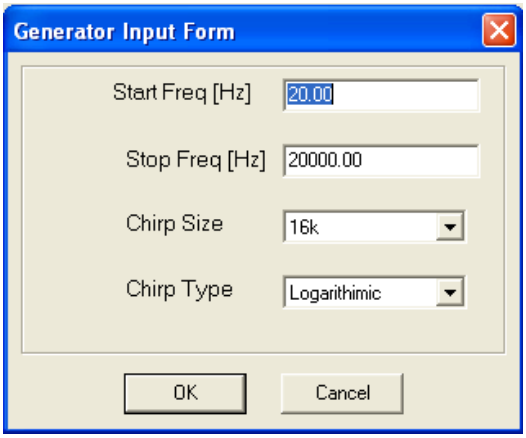
It is possible to generate Chirps (sinusoids with frequency continuously variable with time between two extremes) in two different ways.

You may generate full spectrum Logarithmic Chirps of given length selecting the LogChirp choice in the generator menu.

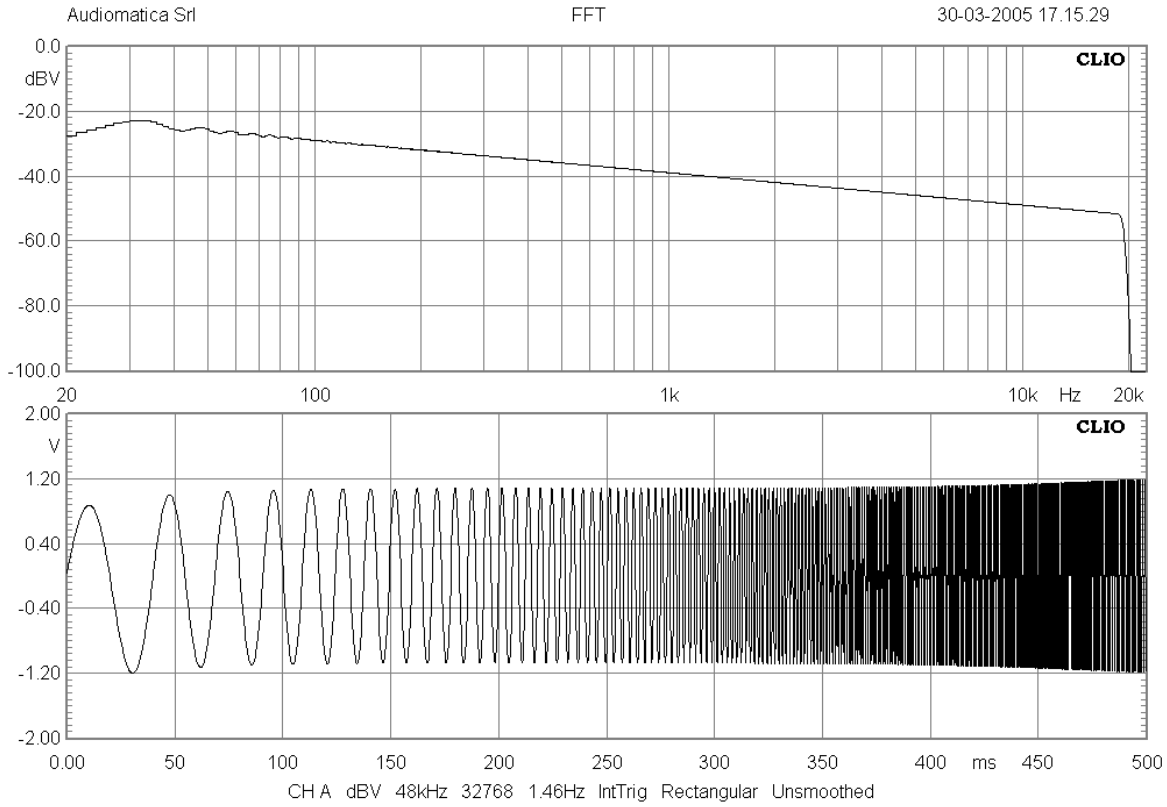


These signals are the same used in the LogChirp analysis menu and should be used to test them.

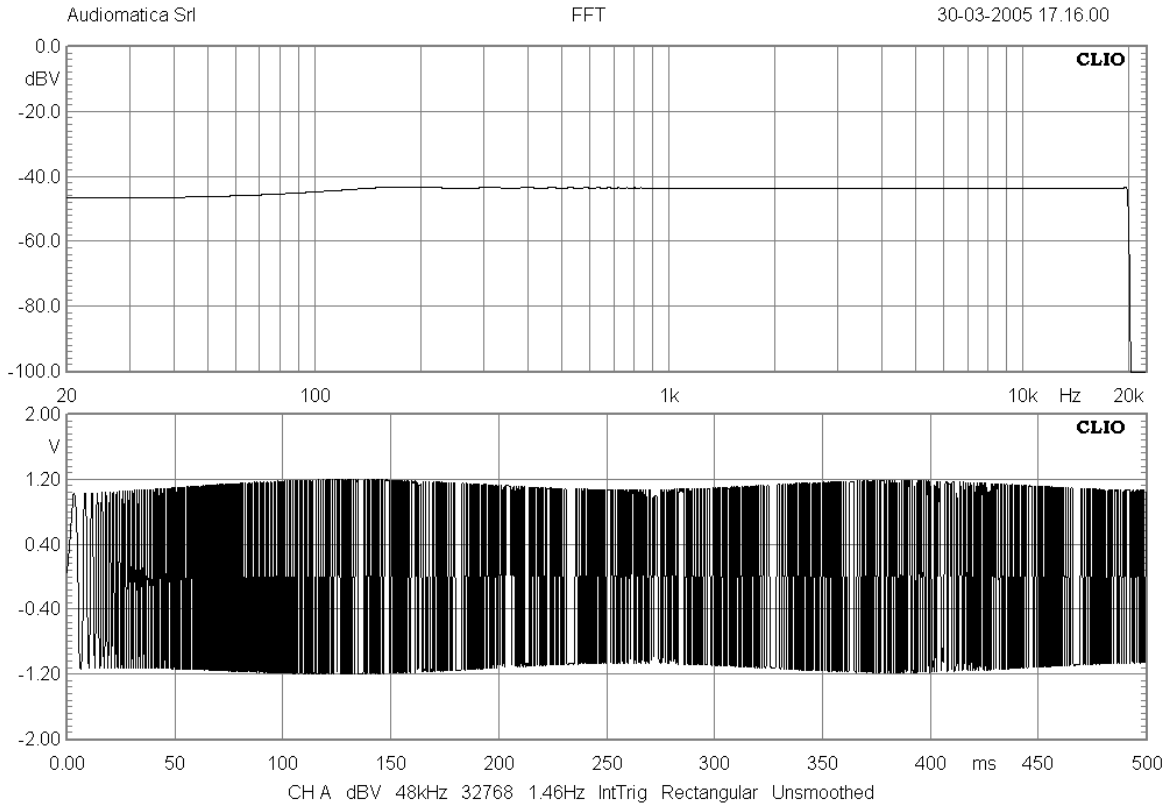
You may instead define Chirps of given length, frequency extremes and kind (linear or logarithmic) selecting the Chirp choice in the generator menu.



The following figure shows a 20Hz to 20 kHz Log Chirp.

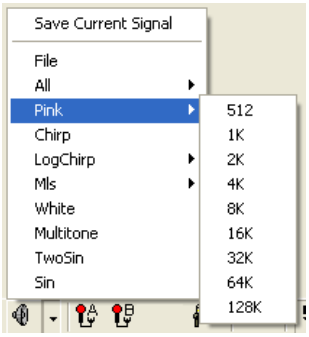


The following figure shows a 20Hz to 20 kHz Lin Chirp.

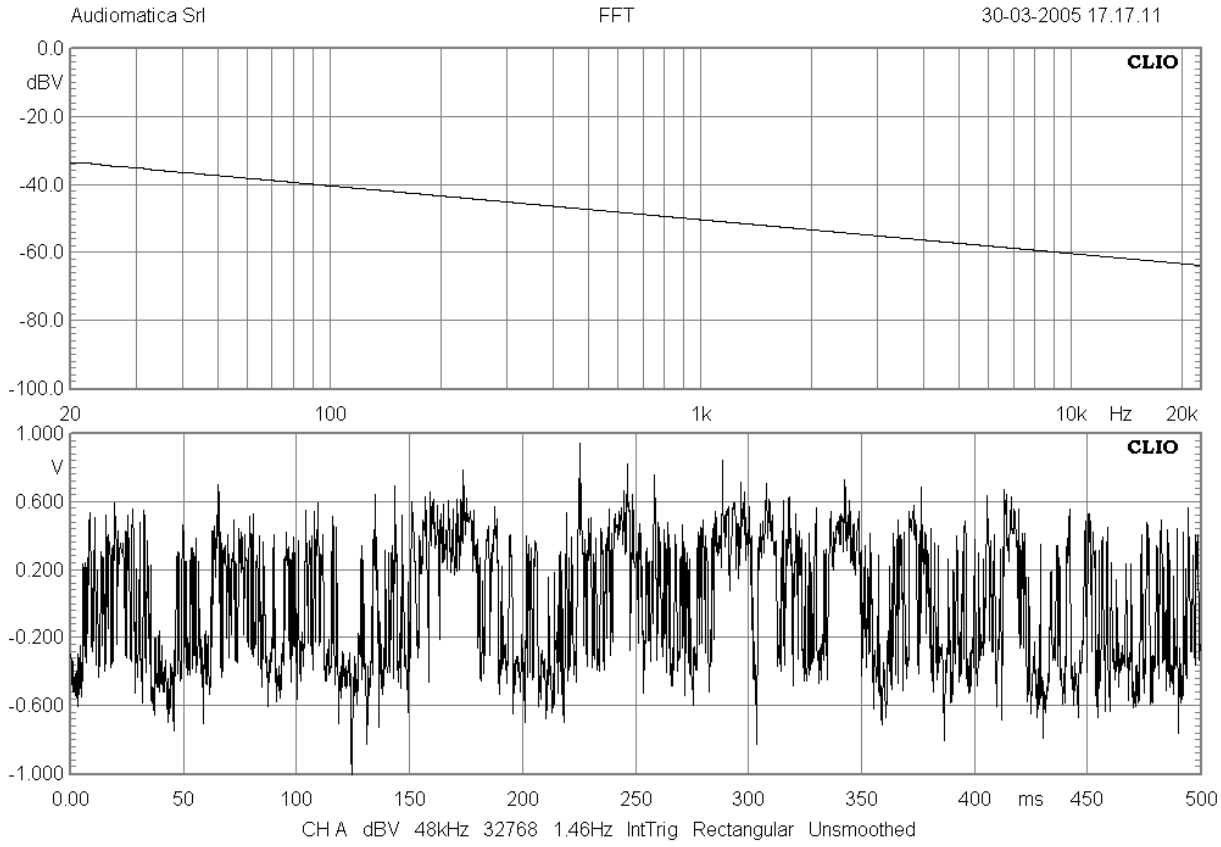


7.8 PINK NOISE

It is possible to generate Pink noises of given length. Select the Pink choice in the generator menu.

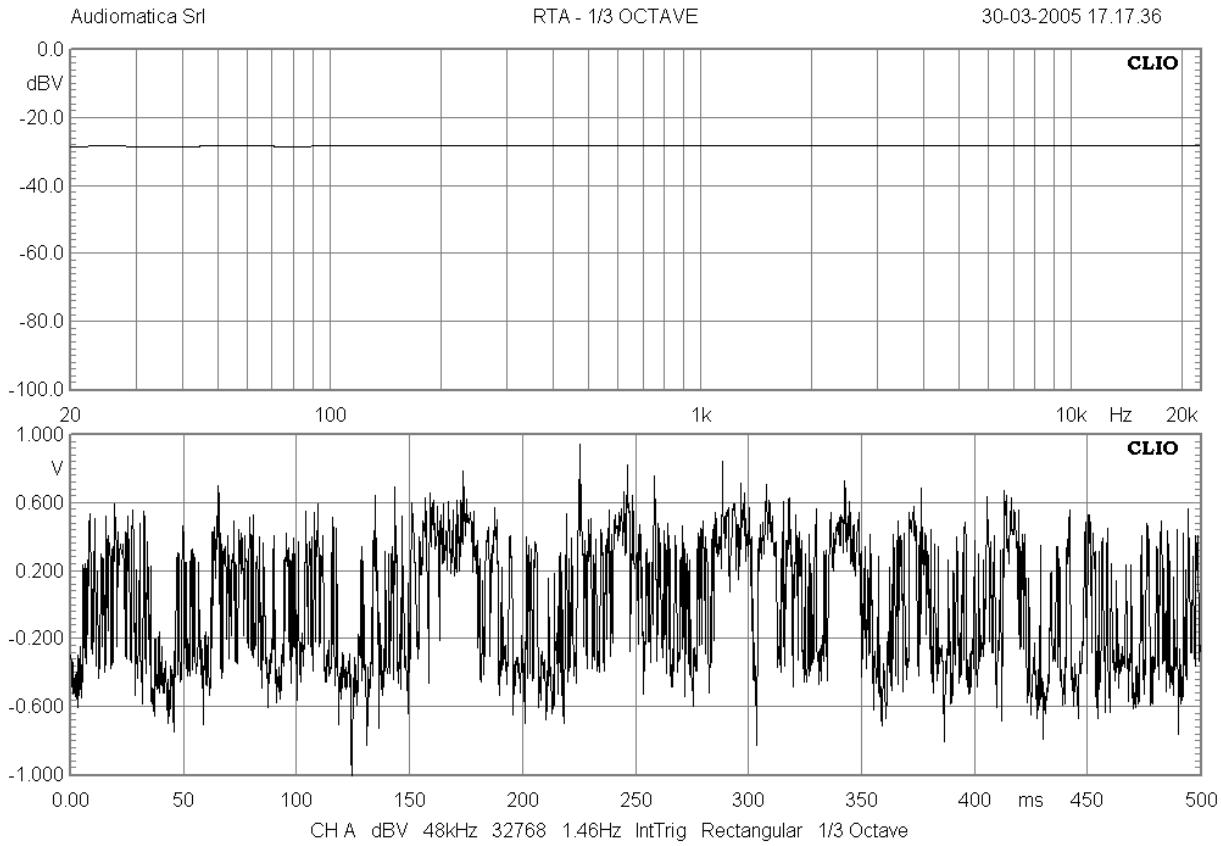


The following figure shows a Pink Noise signal of 32k length measured with the FFT narrowband analyzer.



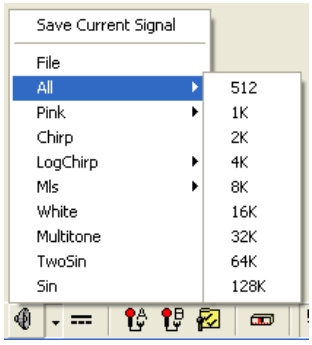
Pink noise signals are used normally to execute Octave bands analysis with the RTA menu due to the flat reponse they produce when analyzed with fraction of octave filters.

The following figure shows the same Pink Noise signal of above measured with the RTA analyzer.

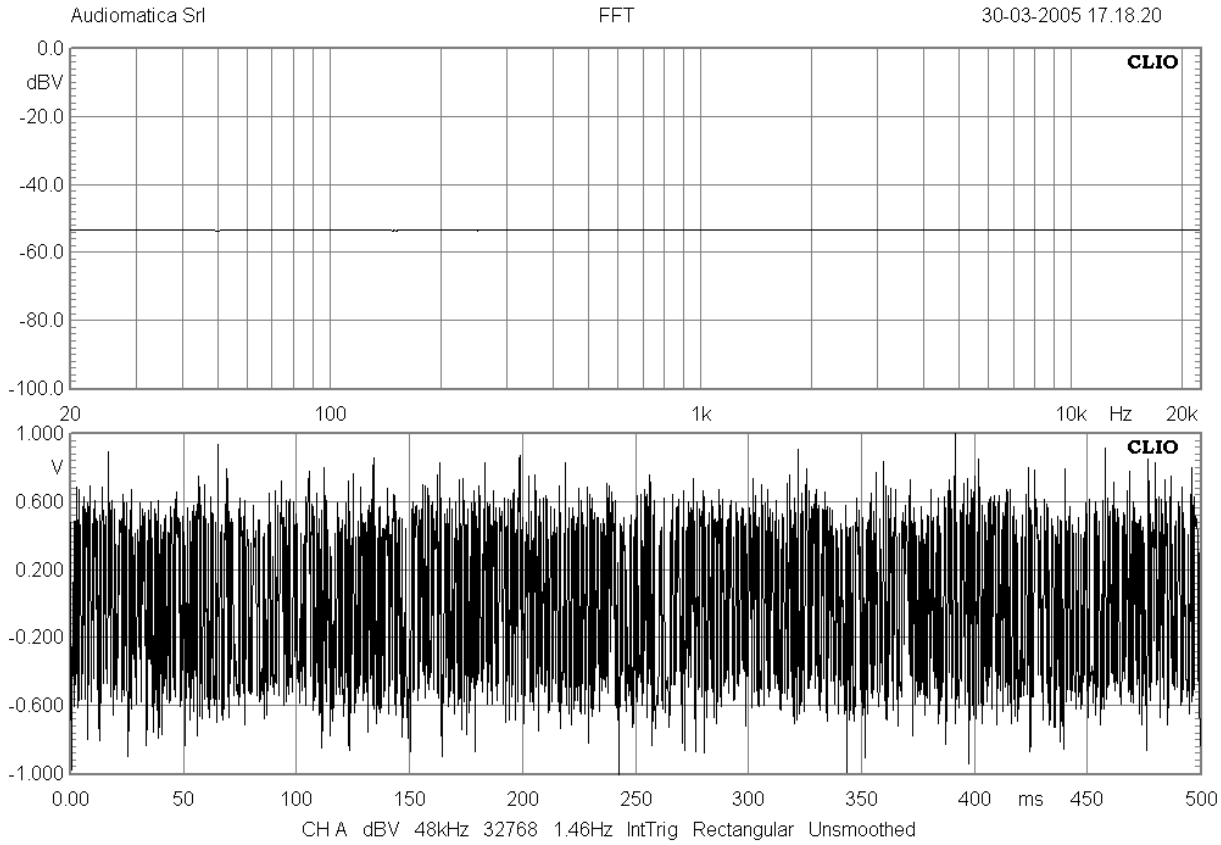


7.9 ALL TONES

It is possible to generate All Tones signals of given length; an All tones contains a sum of sinusoids of frequencies corresponding to each frequency bin with respect to their length and sampling frequency. Select the All choice in the generator menu.

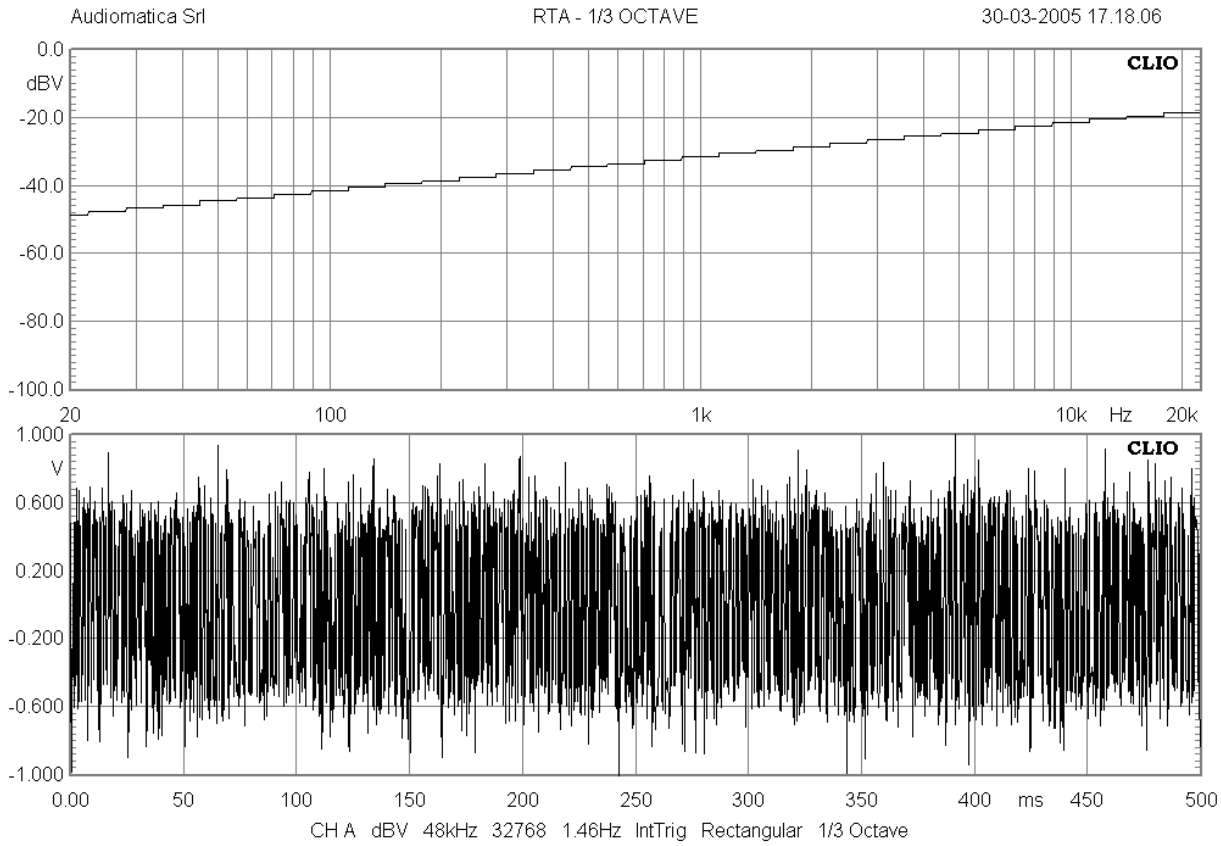


The following figure shows an All Tones signal of 32k length measured with the FFT narrowband analyzer.



All Tones signals are used with the FFT narrowband analyzer due to the flat reponse they produce.

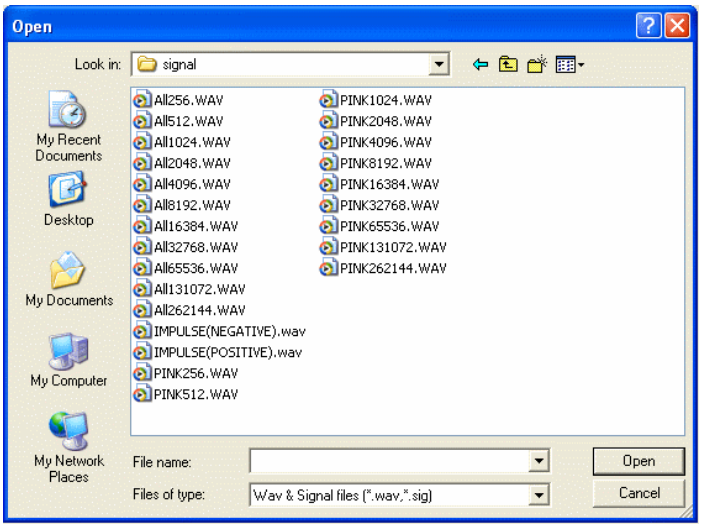
For comparison with Pink noises the following figure shows the same All Tones signal of above measured with the RTA analyzer.



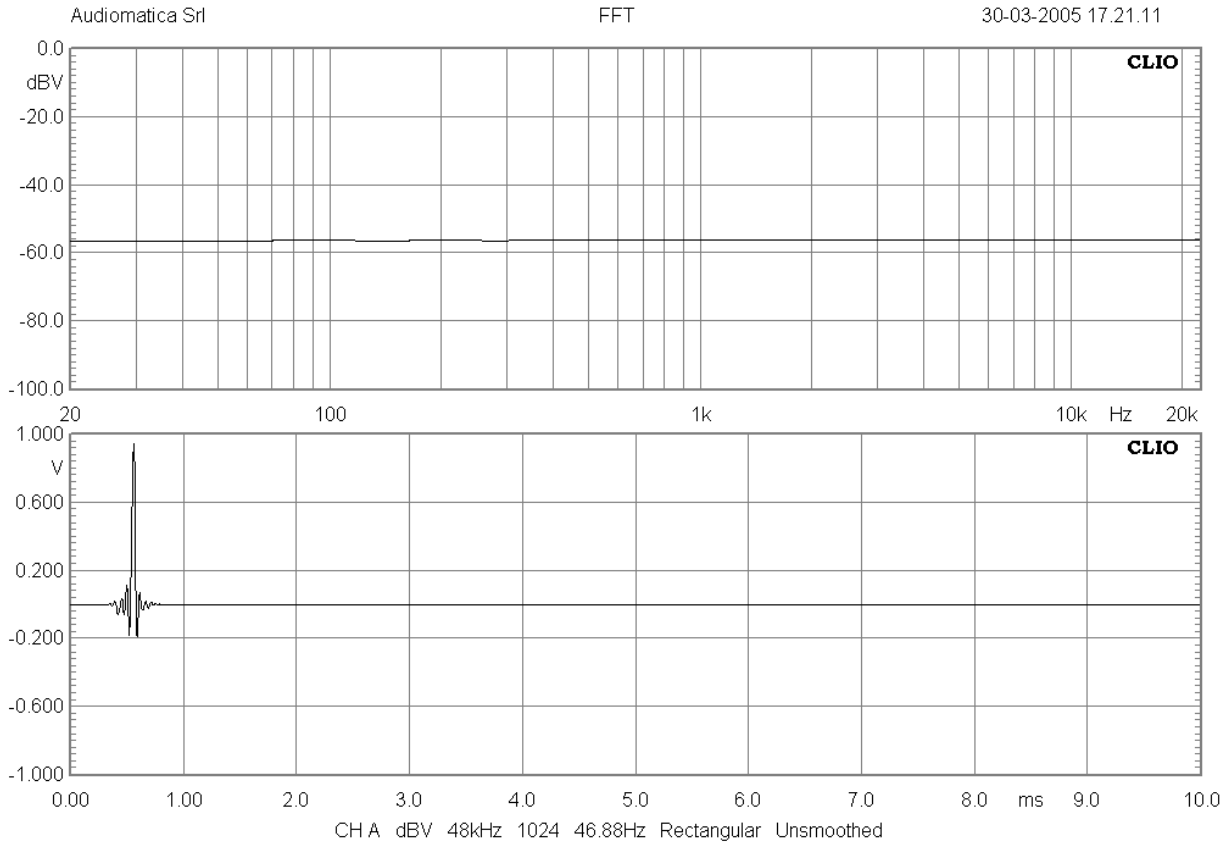
7.10 SIGNAL FILES

As a last possibility it is possible to play signal files saved on disk. Standard **'.wav'** Windows Wave files are supported (**'.sig'** CLIO Signal files are supported for compatibility with older versions of the software).

Choose File within the generator menu. The default extension lets you select a CLIO signal file.



The following figure shows the IMPULSE(POSITIVE).WAV signal file.



The generator menu also keeps track of the **recently** generated signal files giving you instant access to them.

7.10.1 SAVING SIGNAL FILES

The generator menu allows you also to save the current signal present in memory to file. To do this choose Save Current Signal; the format supported is .wav.

Please note that it is possible to generate .wav files from the Leq measurement menu; the data captured during Leq measurements can then be saved to disk and later reproduced with the signal generator.

8 MULTI-METER

8.1 INTRODUCTION

The Multi-meter is an interactive, real-time, measuring instrument.

It gives CLIO the functionality of a:

- **Sound level meter (dBSPL, dBA, dBC)**
- **Millivoltmeter (V, dBV, dBu, dBr)**
- **Laser displacement meter (m, dBmeter)**
- **Laser velocity meter (m/s, dBm/s)**
- **Acceleration meter (m/s², g, dBm/s²)**
- **Frequency counter (Hz)**
- **Distortion meter (THD, THD+N, IMD) (%, dB)**
- **Power meter (W)**
- **L-C-R bridge (H, uF, Ohm)**

Recallable simply pressing **F4**, the Multi-meter has the capability of **capturing the global reference level and the microphone sensitivity**;it is rare that you enter CLIO and don't use the Multi-meter, the information and controls available here are of invaluable importance during the normal operation of the whole instrument.

8.2 MULTI-METER CONTROL PANEL

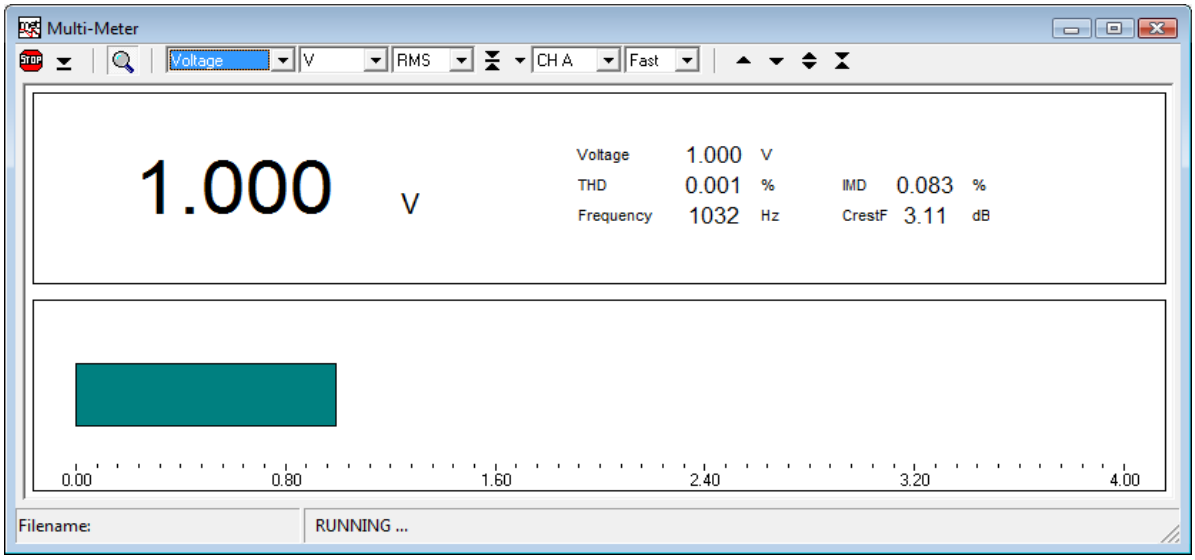


Figure 8.1 Multi-meter control panel

8.2.1 TOOLBAR BUTTONS

Starts the measurement.

 Permits execution with the control panel in a minimized state. Only a small stay-on-the-top display remains visible. See 8.3.2.

Stops the measurement.

If pressed displays all measured parameters.

Captures the actual reading of the multi-meter as the global reference level (or microphone sensitivity); refer to 8.3.2 and 8.4.1 for details.

Control the scale of the meter bar graph.

8.2.2 TOOLBAR DROP DOWN LISTS

parameter
Selects the parameter to be measured.

unit
Selects the measurement unit.

detector
Selects the detector kind among RMS, AVG, Peak, PeakToPeak.

channel
Selects the input channel.

integration
Selects between fast (125ms) and slow (1s) integration. Not applicable to the LCR meter.

8.3 USING THE MULTI-METER

The first application of the Multi-meter has been described in section 3.4.1 when CLIO was started for the first time. This was a simple generation of a 1kHz sinusoid (0dBu output level) and relative level capture with the Multi-meter. You can continue the measurement described to familiarize yourself with the instrument. Pressing the magnifier will let you inspect all the parameters that the Multi-meter measures in parallel (Fig.8.1); changing the selected parameter will bring it to the foreground; for any parameter it is possible to choose different units, for example THD can be shown in percentage or in dB; then you can select the input channel and the integration. This last parameter **(integration)** affects the measurement rate, since the integration **fast** assumes **125ms** of exponential averaging while the integration **slow** assumes **1s** of exponential averaging. The program approximates these constants trying to measure the computer speed and varying the number of averages calculated.

The rest of the paragraph deals with other applications of the Multi-meter.

8.3.1 THE MINIMIZED STATE

It is possible to operate the Multi-meter in what we call a 'minimized' state. Pressing the relative button the control panel collapses and, in its place, remains a **stay-on-the-top** display as in Fig.8.2.

Figure 8.2

This operating mode is very useful; for example suppose you want to measure the total harmonic distortion of a sinusoidal signal while also displaying its frequency content. To do this lets start the FFT and Multi-meter control panels together; then press the Go button in FFT, this will start both measurements (see also 8.6 about this); inside the Multi-meter select the THD parameter and then press the minimized button. You should obtain a measurement situation like the one in Fig.8.3. In this figure we were directly acquiring the output of CLIO while generating a 1kHz sinusoid at 0dBu.

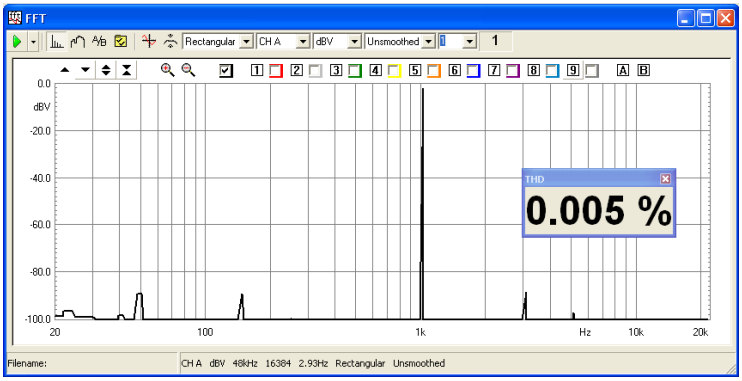


Figure 8.3

8.3.2 CAPTURING THE GLOBAL REFERENCE LEVEL

Now let's see how to capture a level to be the reference for other measurements. Once this is done all subsequent measurements from within the instruments can be referenced to this by choosing **dBREL** as the unit of measurement. As an example,

let us go back to the procedure described in 3.7.1 which aims at validating a calibration. This is substantially the measurement of the frequency response of the CLIO board itself which is, when calibrated, a straight line; as said in the cited procedure the acquired level of such a measurement is -5.2 dBV. Let's see a practical way to acquire this level in order to refer future measurements to it. Keep the instrument connected as in Fig.3.29, with input **A** and output **A** short circuited. Press the MLS button to start generating the MLS signal (the same signal that MLS uses during its operation). Press Go to start acquiring this signal, you should read circa 0.54V. Also select the Slow integration as the MLS signal is a wide band noise. While the measurement is running press the ⧗ button; answering yes to the warning message this will set the global reference level. To inspect the captured value press the button's drop down menu. You should see the situation in Fig.8.4.

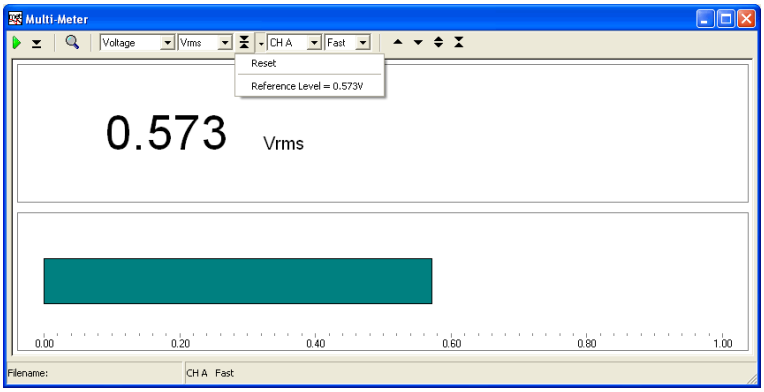


Figure 8.4

The drop down tells you what the actual global reference level is. If you want to restore the default, which is 1V, just choose **Reset**. Having completed this should you choose **dBRel** as your units expect to read 0dBRel!

But let's go to the MLS control panel and see how we can reference frequency response measurements. Open the MLS control panel and simply choose dBRel as units. Then start the measurement. You should obtain the results as shown in Fig. 8.5; a general tendency is always a straight line but now the reading is circa 0dBRel.

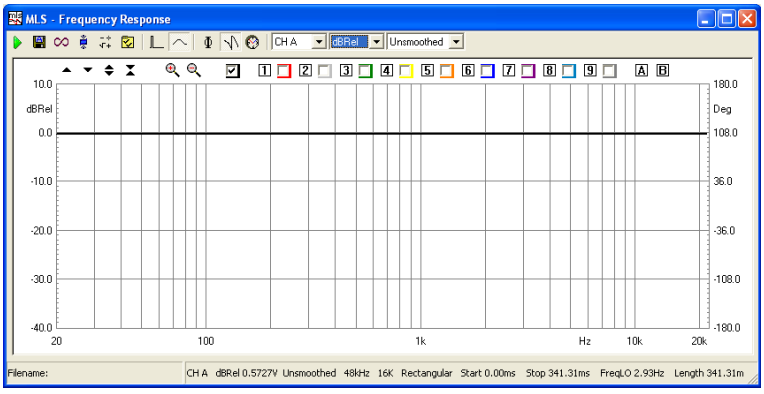


Figure 8.5

8.4 THE SOUND LEVEL METER

Selecting **Pressure** as measured parameter gives your Multi-meter the functionality of a Sound Level Meter. Three units are available: dBSPL, dBA and dBC. **dBSPL** is a direct reading of the sound level, relative to the reference pressure of 20uPa. Remember that CLIO needs to know your microphone sensitivity to carry out this measurement correctly (see 5.4.2). **dBA** and **dBC** are frequency weighted measurements and are usually requested to assess, for example, environmental noise or any human related annoyance. In these two cases the program post processes the measurement, applying the appropriate frequency filter as described in IEC-651 norms.

8.4.1 CAPTURING THE MICROPHONE SENSITIVITY

When measuring pressure, it is possible to calibrate your measuring chain if you have a pressure reference available like an acoustic calibrator. **It is possible to calibrate both channels.**

In this situation, pressing the ⧗ button will prompt the instrument to capture a pressure reading as reference for the reading channel; it expects a reference pressure level at its input, as furnished by the vast majority of acoustic calibrators; it is possible to input its value cliking on the drop down menu; the default value is **94dB**. As a result of this procedure the program will calculate the sensitivity, in mV/Pa, of the microphone front end, and store it in the CLIO settings.

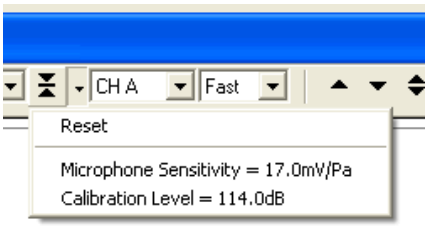


Figure 8.6

If you press the button's drop down you will see a reference to the actual Microphone sensitivity. In this case, differently from the case of Voltage measurements, the default measurement is 17 mV/Pa, which is a rough average of Audiomatica's microphones MIC-01 and MIC-02 sensitivity. Let's see how to capture the channel **A** microphone sensitivity. In Figure 8.7 you see a Bruel & Kjaer 4231 acoustic calibrator fitted to a MIC-02 microphone.

Figure 8.7

With the Multi-meter running, fit the calibrator in place and switch it on. Wait a few seconds for the measurement to stabilize. Then press the ⧗ button. You will receive the prompt in Fig.8.8.

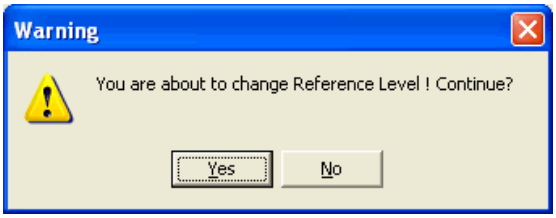


Figure 8.8

Be advised that, by answering yes, you will affect all pressure measurements executed with input channel A. You can inspect the newly acquired sensitivity entering the CLIO Options>Unit Conversion dialog (see 5.4.2).

8.5 THE LCR METER

This is a particular operating mode of the Multi-meter that gives you the possibility of measure inductors, capacitors and resistors. This measurement is an **impedance measurement** and is carried out in the **Internal Mode**; please use Chapter 13 as a reference concerning impedance, related connections and operations.

In this mode the Multi-meter takes control of the generator and, when the measurement is started, outputs a series of sinusoids of variable frequency in order to find the best one to carry out the measurement. The output frequency can be displayed (with the magnifier button) together with the measured parameter.

8.5.1 MEASURING AN INDUCTOR

Measuring an inductor is as easy as connecting it as in Fig.8.9, selecting the In-Out Loop with and press Go. The most critical factor influencing this test are connections; as with all impedance measurements, if you use pin-to-alligators cables use great care in avoiding false contacts which may arise if terminals are oxidized or alligators loose their bite.

Figure 8.9

In a few seconds the measure stabilizes to the final result shown in Fig. 8.10. This was a 4.8mH nominal inductor value; the panel also shows the test frequency that in this case is 502Hz. The same procedure should be carried out when measuring resistors or capacitors.

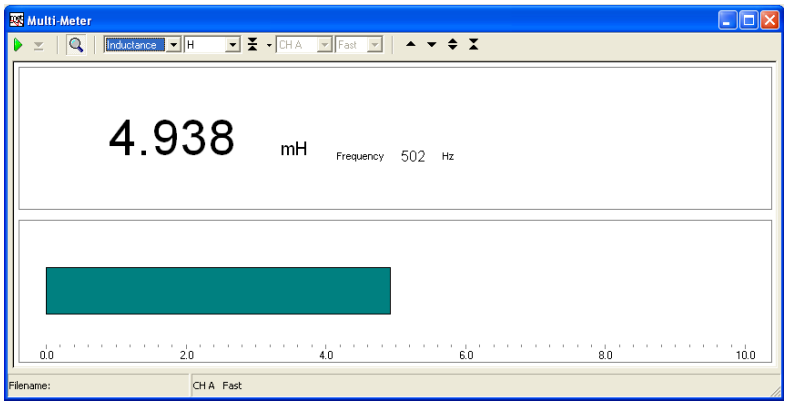


Figure 8.10

8.6 INTERACTION BETWEEN THE MULTI-METER AND FFT

The Multi-meter uses the same capture and processing units as the FFT control panel. To perform a measurement it programs the FFT routines (changing FFT settings to match its needs) and then effectively starts an FFT measurement in background.

The two panels can be opened and can work together but FFT always acts as the **master** while Multi-meter as the **slave**. In this situation the Multi-meter window's title is changed to **'Multi-Meter (FFT slave)'** to reflect this new state; the **Go and Stop buttons are disabled** as you operate the slave panel from inside FFT, starting and stopping the reading with the FFT's Go and Stop buttons; **the input channel follows the FFT one** while **the integration setting become meaningless**; this is because the user has control over FFT averages which precisely define the measurement integration. Nevertheless, during slave operation, it is possible to select the displayed parameter and its unit. It is not possible to select LCR operation as the LCR meter uses different measurement capabilities.

As soon as the FFT control panel is closed it releases Multi-meter from the slave state; then the Multi-meter is then ready to operate in a stand-alone capacity and is fully functional as described above.

9 FFT, RTA AND "LIVE" TRANSFER FUNCTION

9.1 INTRODUCTION

By selecting the FFT command from the main menu bar, it is possible to carry out Fourier analysis of the input signal to determine its frequency content using the Fast Fourier Transform (FFT).

The ability to process two channels simultaneously, to select the appropriate sampling frequency and the possibility of triggering with respect to the generated signal make this control panel a flexible and valuable instrument. Finally, there is also a very useful facility to quickly and easily swap back and forth between the time and frequency domains.

The FFT processed data coming from the two input channels can be displayed as **narrowbands**, **1/3** (or 1/6) **octave bands** (turning the instrument into what is generally called a real-time analyzer or **RTA**) or, referencing one to the other, as **live transfer function** (you may also use music as stimulus).

What you get are three different instruments in one.

9.2 NARROWBAND FFT ANALYZER

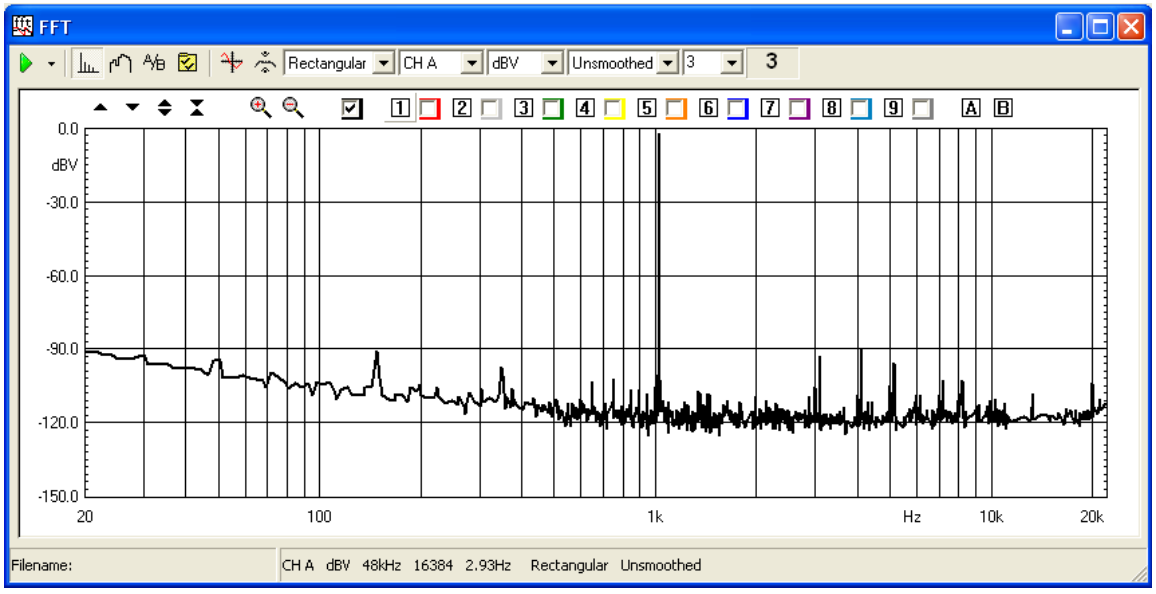


Fig. 9.1 The FFT control panel

Pressing the FFT button you select the narrowband FFT analyzer. Fig. 9.1 shows the FFTcontrol panel (while processing a sinusoidal input signal). The narrowband FFT analyzer is a general purpose instrument that can be applied to the frequency and time analysis of any kind of electroacoustical signal.

There now follows a description of the FFT control panel toolbar and settings.

For a detailed description of the graphical display (common also to other measurement control panels) and its capabilities please refer to Chapter 6. For a description of the available shortcuts please refer to section 4.5.2.

9.2.1 TOOLBAR BUTTONS, DROP DOWN LISTS AND DISPLAYS

gonew.gif Starts an FFT measurement. Right-clicking on it you open the associated drop down menu where it is possible to select the **Continue** switch. In this mode the measurement is not started from blank but accumulates with the previously stopped one; see Averaging (9.6) for details.

Stops the current measurement.

Enters the FFT Settings dialog box.

Enables the Time Data display. The second graph that is activated behaves as an **oscilloscope** and displays the captured waveform correspondent to the last FFT analysis.

Enables the Hold function. Depending on the setting entered in the FFT Settings dialog box, it is possible to hold either the minimum or maximum value per frequency point.

Moves the equal loudness curve up of 1phon. See also 9.5.

Moves the equal loudness curve down of 1phon. See also 9.5.

data window

Selects a weighting data window among the following:

- Rectangular (no window)
- Hanning
- Hamming
- Blackman
- Bartlett (triangular)
- FlatTop

channel display

Selects the input channel to display among the following:

- Channel A only
- Channel B only
- Channel A&B (both channels displayed in different graphs)

Y scale units

Selects the measurement units among the following:

- dBV
- dBu
- dBRel (with respect to the global reference level; see chapter 8)
- dBSPL (for acoustical measurements)

frequency smoothing

Selects the frequency smoothing in fractions of octave from 1/48 to 1/2 of octave.

target averages

Inputs the total number of averages. Averaging is controlled by the setting in the FFT Settings dialog; see Averaging (9.6) for details.

number of averages display

Displays the number of the actual average; this number increases during the

measurement unless, while in exponential averaging, the target has already been reached; see Averaging (9.6) for details.

9.3 RTA - OCTAVE BANDS ANALYZER

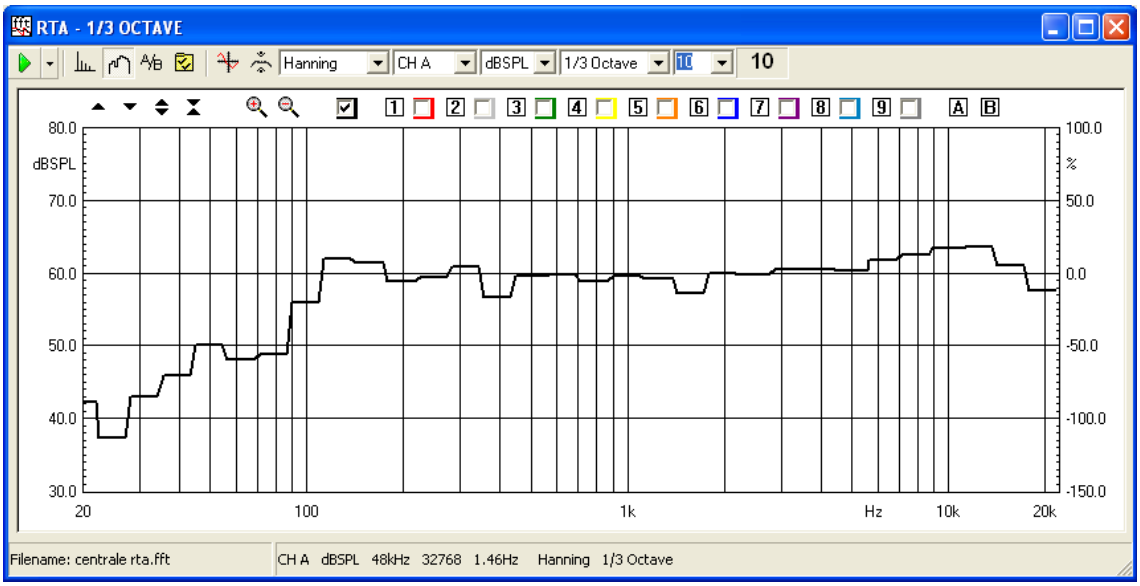


Fig. 9.2 The RTA control panel

Pressing the RTA button you select the octave bands analyzer. Fig. 9.2 shows the RTA control panel (while analyzing the 1/3 octave response of a HT center channel speaker). The RTA (RT stands for real-time) analyzer is a dedicated instrument that is normally used to measure the averaged frequency content of an acoustical signal and for evaluating how a complete reproduction system (from PA to HT) behaves in a real environment.

For a detailed description of the graphical display (common also to other measurement control panels) and its capabilities please refer to Chapter 7. For a description of the available shortcuts please refer to section 5.5.2.

9.3.1 DEDICATED TOOLBAR FUNCTIONS

fraction of octave

Selects between 1/3 and 1/6 of octave analysis.
This is the only different toolbar function from the FFT narrowband case; for all the other functions refer to 9.2.1.

9.4 FFT SETTINGS DIALOG

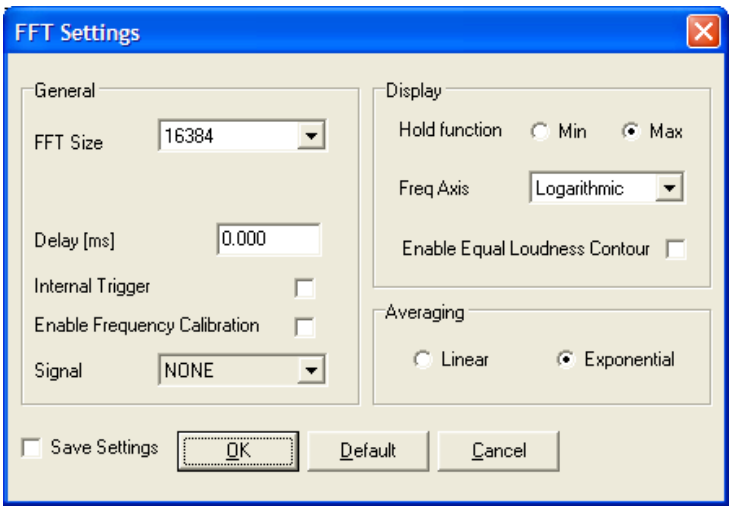


Fig. 9.2 The FFT settings dialog box

FFT Size

Selects the number of samples acquired and processed by each FFT. It is possible to choose a size between 512 and 131072 points.

Delay

Permits the input of the desired processing delay (in ms) when in Internal Trigger mode. See 9.7 for details.

Internal Trigger

Enables the Internal Trigger mode. See 9.4 for details.

Enable Frequency Calibration

Enables frequency calibration to compensate for any hardware non linearity; frequency calibration, if enabled, takes place only when the generator is active.

Hold Function

Selects either Min or Max hold function. This functionality is activated by the relative toolbar button.

Freq Axis

Selects from linear or logarithmic frequency axis (valid only for FFT narrowband)

Enable Equal Loudness Contour

Enables the display of the normal equal loudness level curves as defined in the ISO 226 standard. The curves are displayed only in FFT narrowband and RTA modes when dBSPL units are selected.

Averaging

Selects either linear or logarithmic averaging; see Averaging (9.6) for details.

9.5 FFT AND RTA OPERATION

The FFT and RTA measurements (and also Multi-meter ones, see Chapter 8) differ from MLS and Sinusoidal ones in the fact that they are interactive; the user has control over measurement time and generated stimuli. You may also obtain answers about unknown signals from them, without any need for generating a stimulus; or you may leave this job to others, similar to when you measure an audio chain relying on the test signals contained in a CD-ROM. One effect of this is that, strictly speaking, FFT measurements may lead to less precise results if compared to other techniques; the possibility of injecting a synchronous MLS sequence at the beginning of the same audio chain mentioned before is surely a better approach even if, in the vast majority of cases, unfeasible.

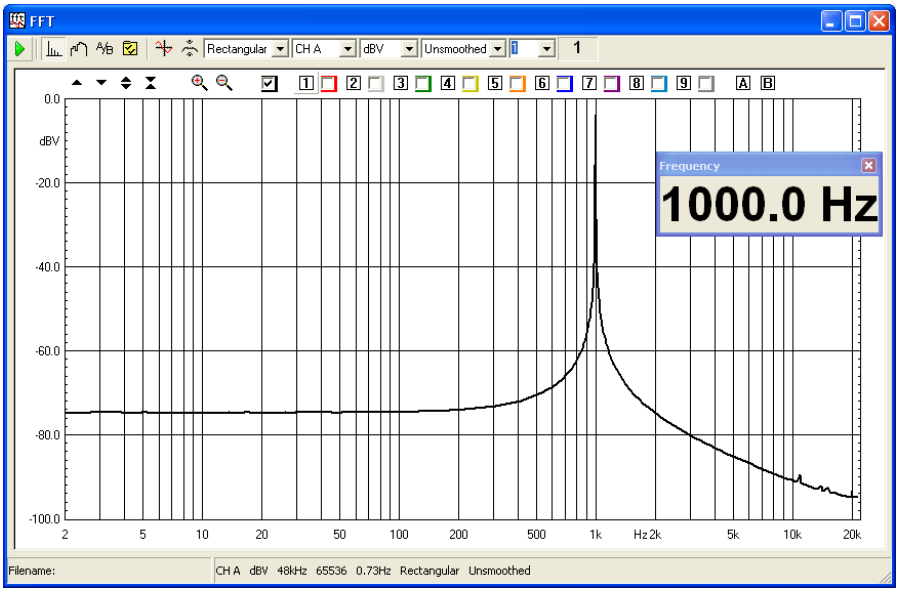
FFT and RTA power depends not only on the measurements settings themselves but also on the **generated signals**. Please refer to chapter 7 for a detailed description of the signal generator and its many capabilities.

When stimulating any external device with CLIO (see 4.8.2 and 4.8.3 for basic connections) you may choose a limited bandwidth signal (like a single sinusoid) or a wide bandwidth signal as a noise; in the first case you have the possibility of analyzing the harmonic content of the output spectrum while in the second case you may evaluate the frequency response of the device under test. A different stimulus, about halfway between the two cases just mentioned, is a logarithmic chirp swept across some octaves (like a chirp covering four octaves from 50 to 800Hz); in this case you are able to analyze both the response plus unwanted effects like distortion and noise produced by the D.U.T..

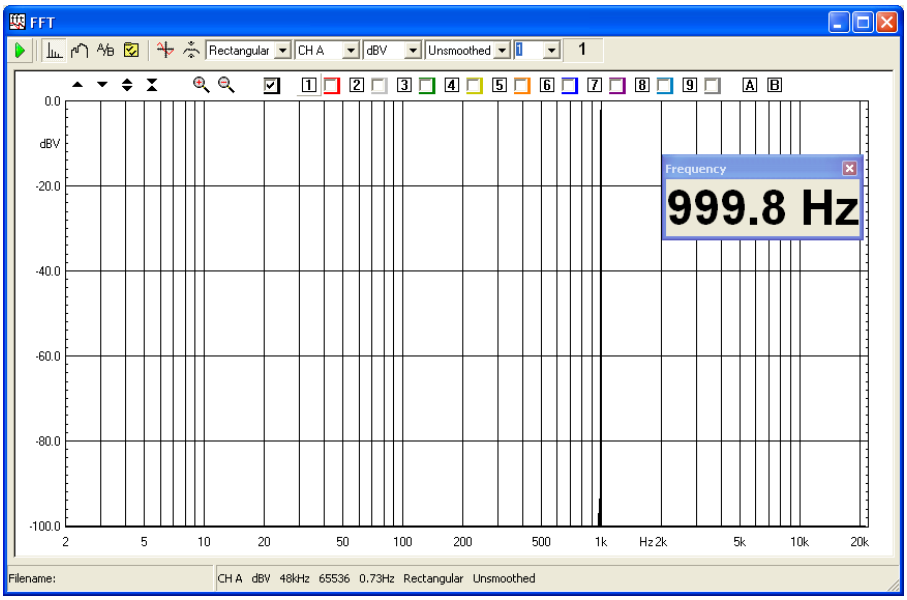
When using the FFT narrowband analyzer it is possible to achieve a flat response of the analyzing chain using white noise or similar signals whose energy content varies **linearly with frequency**; among these MLS, All-tones signals or linear Chirps.
When using the RTA octave bands analyzer it is possible to achieve a flat response of the analyzing chain using a signal whose energy content varies **logarithmically with frequency**; among these we find pink noises or logarithmic Chirps.

Besides the choice of the stimulus it is very important to achieve proper **synchronization** between the generated signal and the acquisition; this will lead to optimum performances avoiding the use of data windows and minimizing any spectral leakage that may occur. Synchronization can be achieved defining the stimulus in a particular manner or by means of proper triggering (see later **internal trigger**).

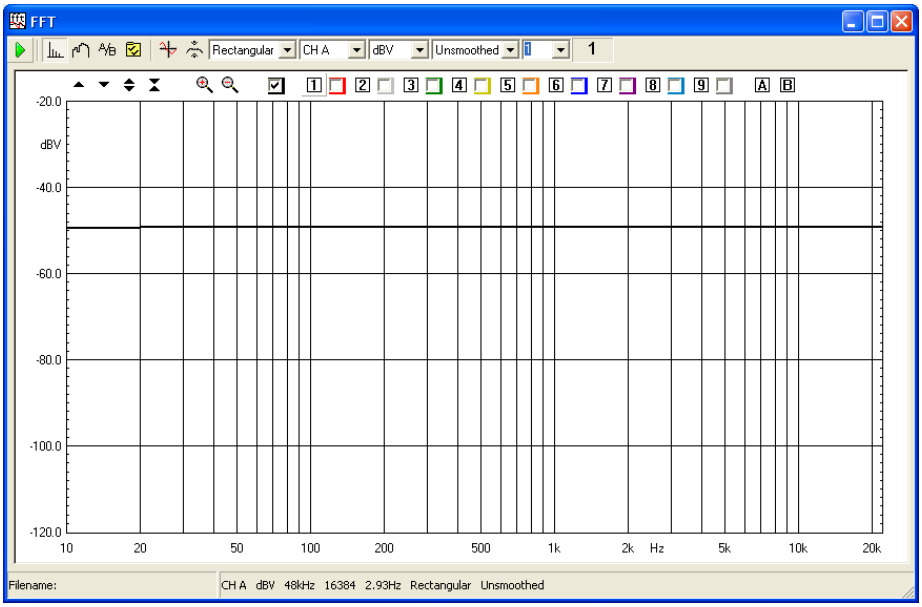
If you are generating a **sinusoid** choose a frequency that is an integer multiple of the frequency **bin** (i.e. sampling frequency divided FFT size) or let CLIO calculate it setting "FFT bin round" in the generator input form (see chapter 7). As an example we would like to play and analyze a 1kHz sinusoid using a 64k FFT @ 48000Hz sampling; the frequency bin associated is 0.73Hz and the nearest spectral line to 1kHz is the 1365th one at 999.75Hz. If you simply generate a 1kHz sinusoid without rounding it to the nearest bin you obtain the analysis of fig. where it is evident that CLIO is capable of outputting a highly precise 1000.0Hz sinusoid but it is also evident the spectral leakage caused by this choice.



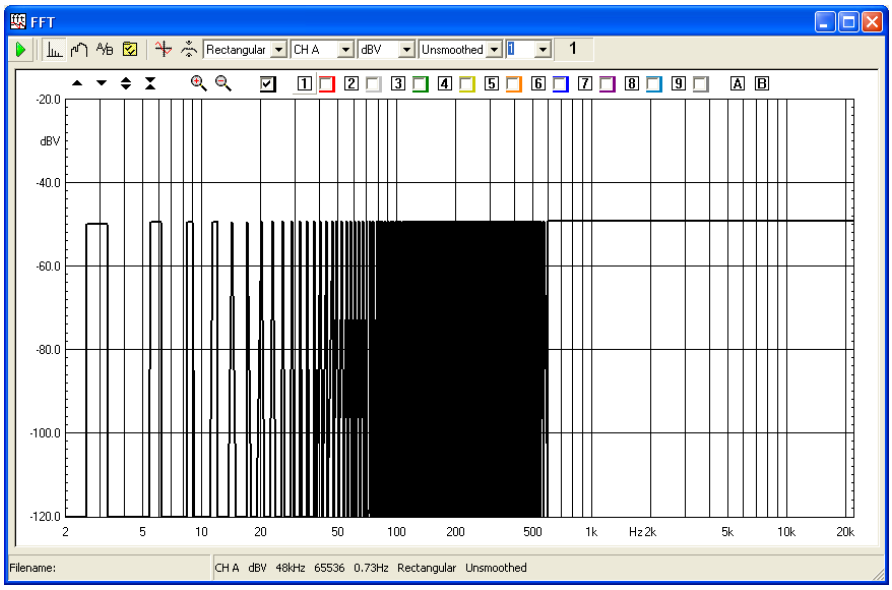
A better approach is to center the sinusoid to the nearest spectral line i.e. 999.75Hz as shown in the next figure. Note the use of the multimeter as frequency counter; note also that its precision is of 0.1Hz when FFT size is higher than 32k.



If you want to generate a full spectrum signal choose an All-tone of proper length matching FFT size. The following figure shows a 16k All-tone (all16384.sig) analyzed with a 16k FFT.



If you had chosen a wrong size, like an all tone of 8k, you would have obtained the following analysis which clearly shows a lack of energy at alternating bins; the effect is visible only at low frequency due to the logarithmic nature of the graph.



CLIO has the possibility of **internal trigger** (and relative **delay**) i.e. triggering with respect of the generated signal thus obtaining a synchronous capture. As an example let's see how a measurement presented in section 11.4 was done; please refer to figures 11.9, 11.10 and 11.11. We have an acoustical measurement of a tweeter, done stimulating it with a 2kHz 10ms tone burst (see 5.4.2 for details about programming a bursted sinusoid); the FFT measurement is done using the internal trigger; Fig. 11.9 shows the analysis and the captured time data that clearly shows the flight time from the tweeter to the microphone, Fig. 11.9, Even if the analysis is not our final target, it shows the power of synchronous acquisition which permits the display of the arrival delay of sound to the microphone. To obtain the desired result, as explained in 11.4, it is necessary to remove the flight time plus the device settling time; this can be easily accomplished setting the internal trigger delay, in FFT settings, to 1.5ms; the final result shown is shown in 11.11

and permits the identification of the device harmonic distortion. To proceed further one could vary the stimulus amplitude and test the distortion of the tweeter at different amplitudes; using bursts also prevents the damage of the unit as the overall power delivered to it rather low and a direct function of the duty cycle of the burst itself.

The main application of RTA analysis is in assessing the quality of an audio installation (from the placement of the speakers in a listening room to the overall sound quality of a car stereo system). In these cases pink noise is often used as the stimulus. If you are not using CLIO as the source of such a stimulus be sure to use a good one; you may find several audio generators that do the job, but they are usually expensive. A good choice is to use a recorded track of one of the various test CDs available; in this case not all the CD-ROM readers may furnish adequate results, as appears from the graph in Fig.9.3

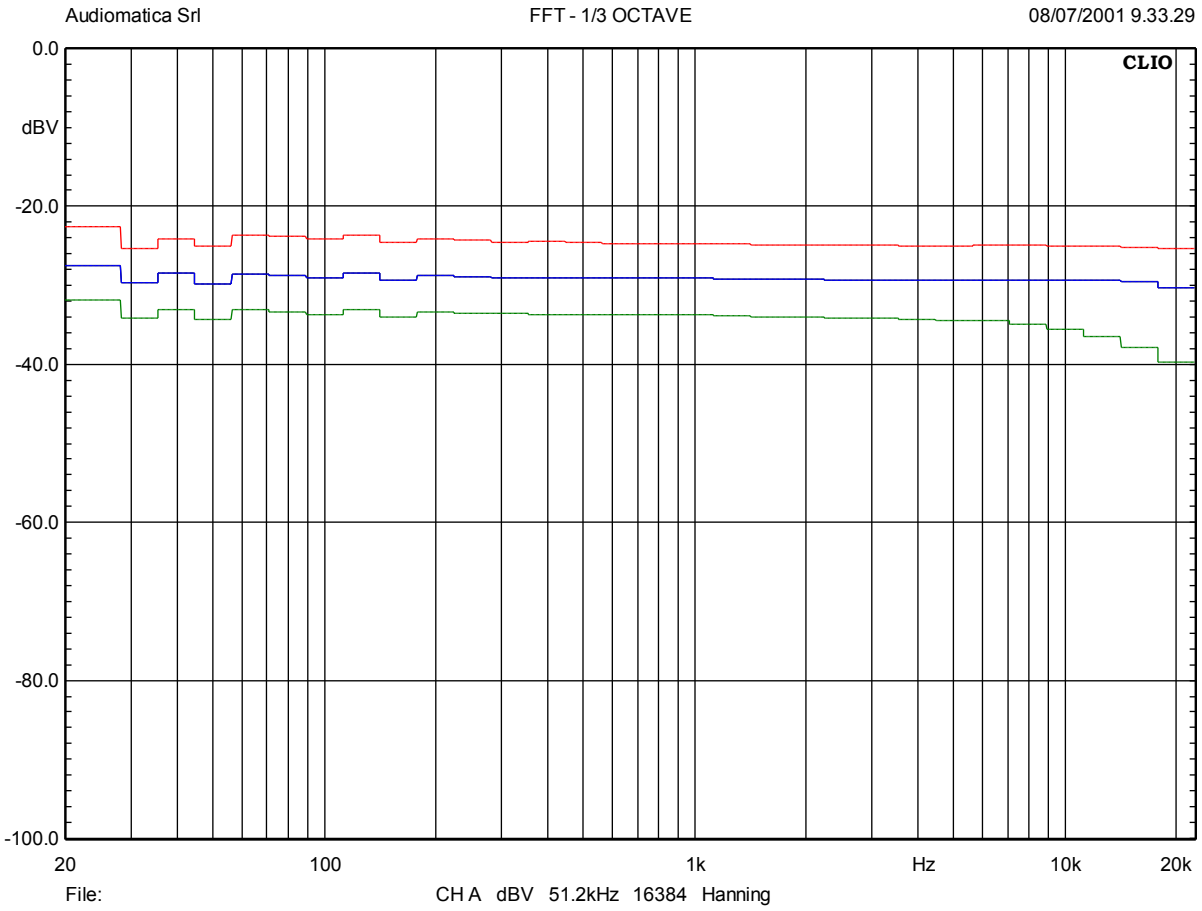


Figure 9.3

All three graphs represent true analog pink noise, they are played at intervals of 5dB for clarity. The upper (red) is the output of an Audio Precision System One generator; the second (blue) is the pink noise of track 4 of the Stereophile Test CD played by a Philips CD692 CD player, the third is the same track of the same test CD output by the computer which I'm writing with right now (Pioneer DVD Player plus Crystal Sound Fusion PCI Audio).

When taking RTA measurements use, at least, 16k FFT size if you want to cover the entire 20-20kHz audio band; using lower sizes results in octave bands not present as no FFT bins fall inside them.

9.6 AVERAGING

Averaging plays a very important role in FFT analysis. It is vital when analyzing signals buried with noise. It is also important when taking spatially averaged measurements. CLIO has flexible averaging capabilities. Averaging basically means adding and dividing for the number of additions made. To start an averaged FFT measurement you need to set a number bigger than one in the Target Average drop down; otherwise you have a continuously refreshing single measurement that will continue until the Stop button is pressed. It is possible to choose between **linear** and **exponential** averaging. The instrument behaves differently in the two averaging modes. In linear averaging the measurement is continued until the target is reached, then it automatically stops. What you get is exactly what we have just explained. In exponential averaging the measurement never stops. When the target is reached the averaging continues relying on a mathematical formula which discards the oldest acquisitions and gives more and more importance to newer ones. The exponential averaging is the default one.

As an example Fig. 9.4 compares a single 1kHz sinusoid FFT analysis with a 100 averages one.

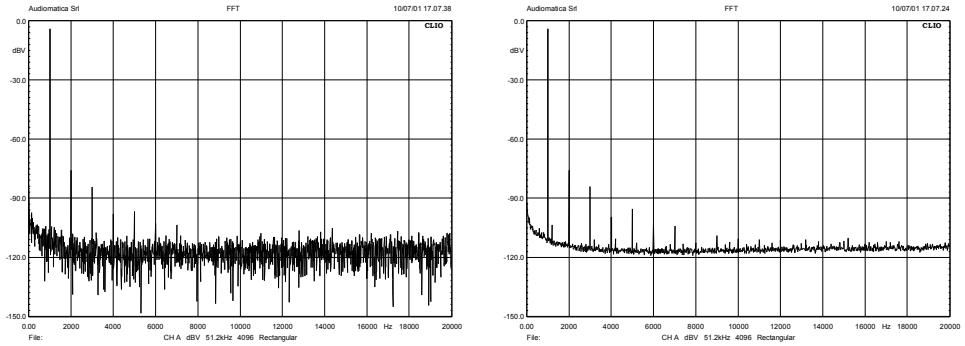


Figure 9.4

This is a classical example of signal buried with noise: the sinusoid's 9th harmonic is clearly visible after 100 averages, but invisible for a single acquisition.

Another important feature, when averaging, is the possibility of **continuing** averaging after a measurement is stopped, has reached its target or a previous measurement has been loaded from disk. CLIO has this capability when selecting «continue» in the drop down menu available beside the Go button (Fig.9.5).

Figure 9.5

Selecting the continue option allows for example, for spatially averaged measurements. Fig. 9.6 shows two 1/3 octave RTA measurements of a small HT satellite at listening position: the black one is a single 10 averages measurement taken on axis; the red one is, instead, built using the continue option, adding a total of eleven 10 averages measurements taken moving from -25 degrees left to +25 right of the speaker itself.

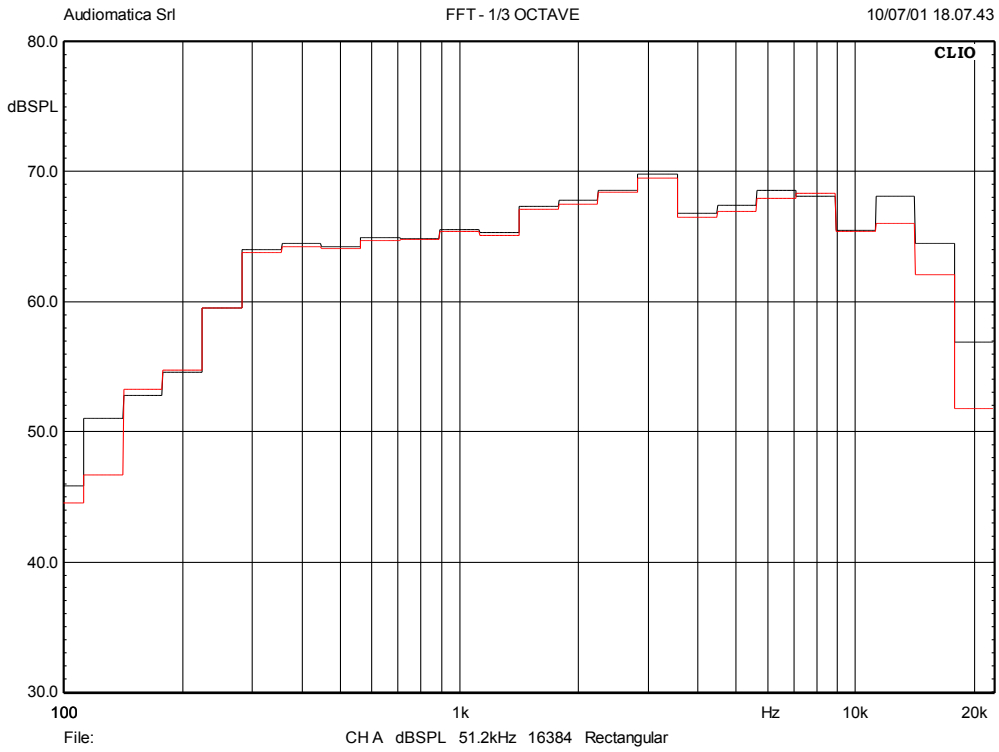


Figure 9.6

9.7 TIME DATA DISPLAY (OSCILLOSCOPE)

The time data (Fig. 9.7) is an ancillary display to an FFT or RTA executed measurement. Here we see a 100Hz sinusoid captured and analyzed with a 16K FFT.

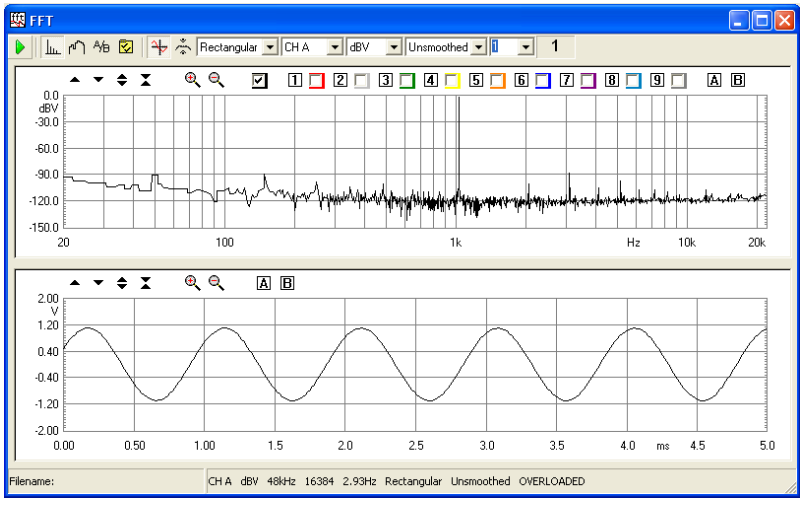


Figure 9.7

The time data display has a number of particular features that allow, **out of user control**, for automatic scale adjustment and triggering of the displayed signal.

The time information displayed is the processed data only. The user can zoom in and out of this data but it is not possible to display more than one acquisition area. Fig. 9.8 shows this effect; the same 100Hz sinusoid as before seems now truncated; this means that we only reached the maximum displayable data, in case of a 16K FFT @ 51200Hz sampling this is exactly 320ms.

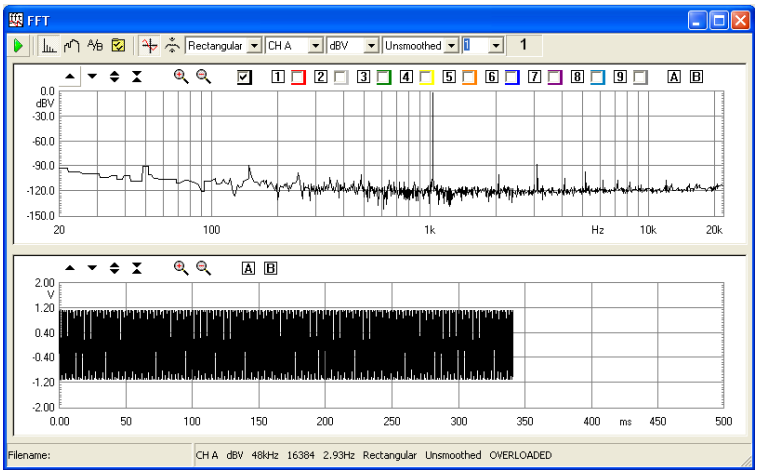


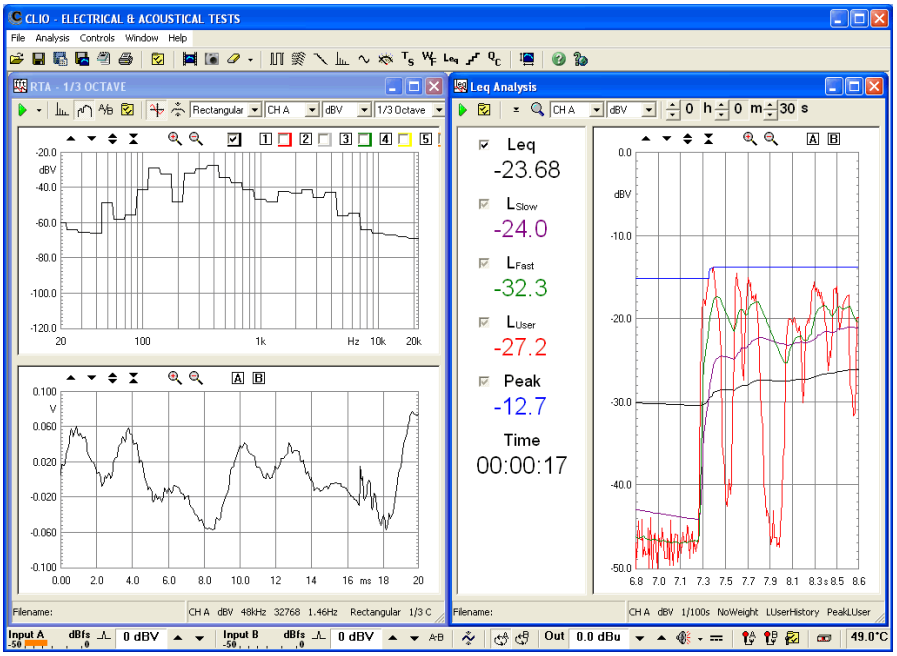
Figure 9.8

9.8 FFT AND MULTI-METER

There is a close interaction between FFT and Multi-meter operations. The two measurements share the same acquisition and processing core. Should they operate together the FFT control panel acts as the **master** while Multi-meter follows as the **slave.** In this situation, among other peculiarities, the Go and Stop buttons of the Multi-meter are disabled; if an FFT acquisition is started then the Multi-meter runs as well, the same when you stop the measurement. More on this is in section 8.6.

9.9 FFT AND Leq ANALIZER

It is possible to execute FFT or RTA analysis while an Leq measurement is taking place. The following figure shows the analysis of a sample of speech done with Leq and RTA in parallel.



Some limitations apply to FFT settings, in particular it is not possible to select an FFT size higher than 32768. See also chapter 16.

9.10 "LIVE" TRANSFER FUNCTION ANALYZER

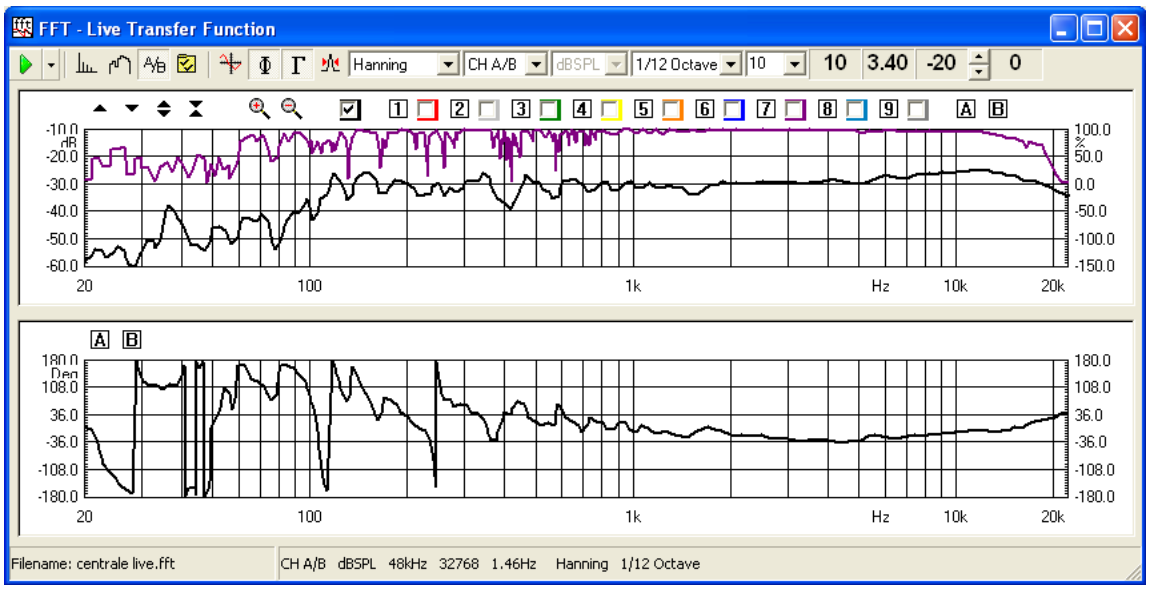


Fig. 9.3 The Live transfer function control panel

Pressing the Live transfer function button the instrument behaves as a dual channel FFT analyzer referencing one channel to the other and calculating the transfer function between the two. Fig. 9.3 shows the Live transfer function control panel (while measuring the frequency and phase response of a loudspeaker). The Live transfer function analyzer is a the fastest way to measure frequency response and has been optimized for interactive and fast refreshing live situations when the only available signal is reproduced music.

For a detailed description of the graphical display (common also to other measurement control panels) and its capabilities please refer to Chapter 7. For a description of the available shortcuts please refer to section 5.5.2.

9.10.1 DEDICATED 'LIVE' TOOLBAR FUNCTIONS

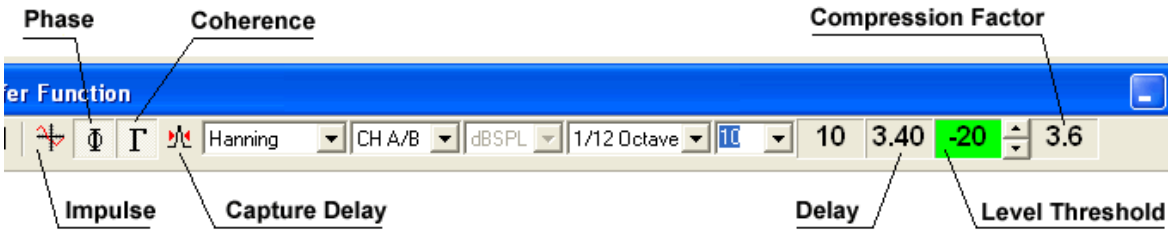


Fig. 9.4 The Live transfer function toolbar

Enables the Time Data display. The lower graph displayed is the **impulse response**.

Enables **phase** response to be displayed in the lower graph.

Enables **coherence** response. The coherence is displayed in the same graph of the transfer function magnitude and reads on the right displays scale.

Automatically evaluates the interchannel delay and sets it. The value of the delay is shown in the delay display.

delay display

Shows the delay correction, in ms, that is applied while processing the two channels.

level threshold display and control

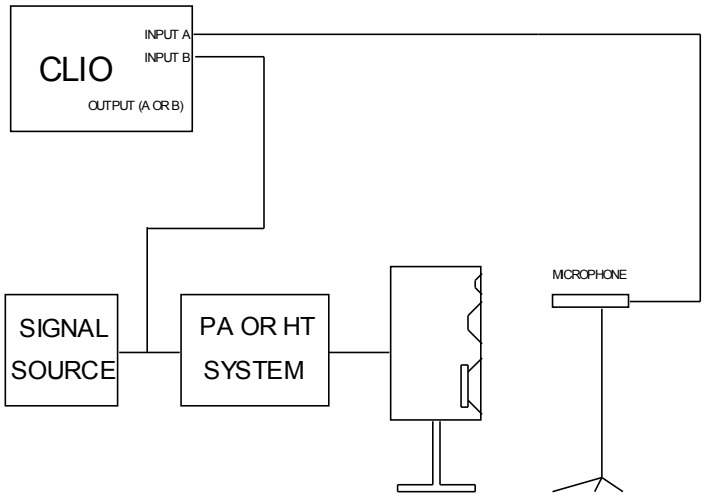
Sets the peak level versus input full scale of the reference channel below which the measurement is frozen. It is possible to modify the value using the dedicated spin buttons. Setting this threshold properly lets you measure only when the signal is present at the reference channel and avoid that inaccurate readings accumulates with the measure distorting it.

compression factor display

Shows the difference, in dB, between the crest factors of the two input signals. The compression factor gives you a rough indication about how much the system under test is limited in its dynamic range. The more negative the compression factor more the system is limiting the input signal.

9.10.2 USING CLIO DURING A LIVE PERFORMANCE

When using CLIO as a dual channel FFT analyzer for measuring a "Live" transfer function you should follow a particular connection diagram which is depicted in figure.



In this case the signal source may be music or any live program material; to monitor the system transfer function you must feed the original signal plus the measured one to CLIO. The input channel to which you connect the original signal is said to be the **reference** channel; in figure it is channel B; you will then measure the A/B transfer function.

Before starting a live measurement choose the proper FFT settings based on your experience; for your first tests we suggest the following:

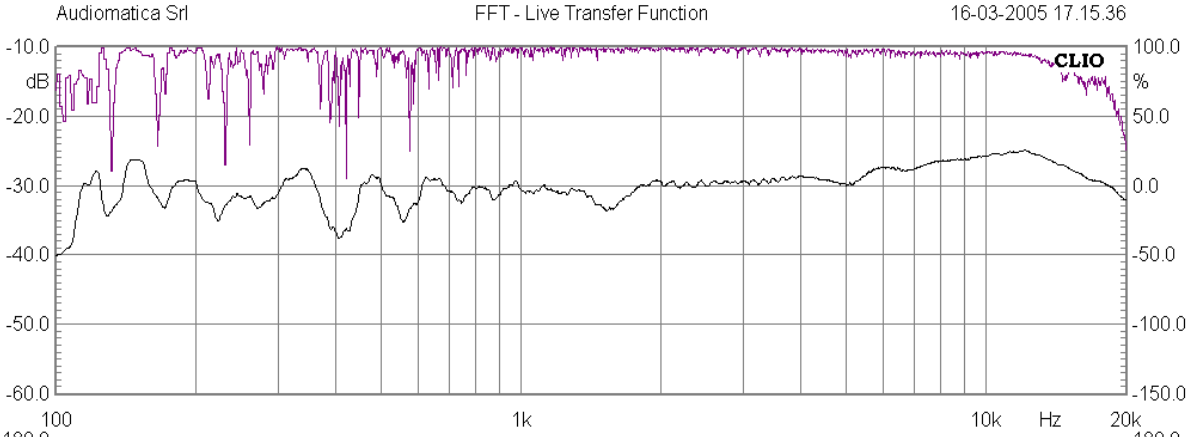
- FFT Size: 32k or more
- Window: Hanning
- Smoothing: 1/48th Octave
- Averages: at least 10
- Level Threshold: -20dB.

Another factor of maximum importance in order to obtain the best results is to properly set input sensitivity for **both** input channels **separately**; the two peak meters of CLIO desktop should help you in this task; set input sensitivity so that both readings average in their respective green areas. If you are taking as reference the line level signal fed to the sound reinforcement system and are measuring from a microphone it is quite common the situation in the following figure where the two input channels have sensitivity which differ of 40dB or more.

When the measurement is started you will see the A/B **magnitude** frequency response which is averaging on the screen and fast refreshing; sometimes you may notice, in dependance of moments when sound is particularly low or absent, that the measurement freezes and the Level Threshold display becomes red:

You may freely choose the most appropriate level threshold so that your measurement accumulates only when signal is present; take into consideration that the threshold relates to peak level vs. full input scale so it is not an absolute value but can be directly compared with the CLIO desktop peak meters.

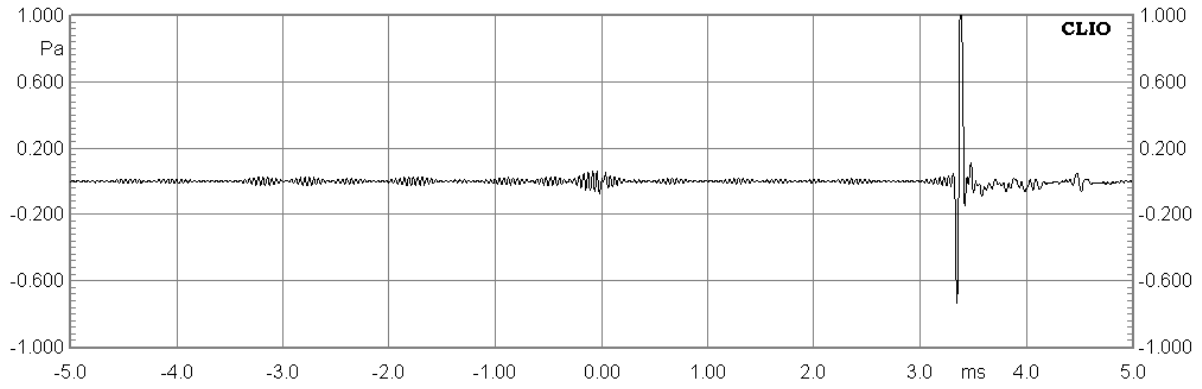
Another important indication about how good is your measurement is given by the **coherence** graph.



The coherence function tells us in which frequency zones the energy content of the measured signal relates, i.e. is coherent, with the reference signal; the more the signals are coherent the more the coherence function approximates unity (or 100%); the less they are coherent the function tends to zero. In figure above we may say that both in the highest frequencies region (above 15kHz) and in the mid-bass region there are zones of low correlation (<50%) where the measured response should be considered with attention.

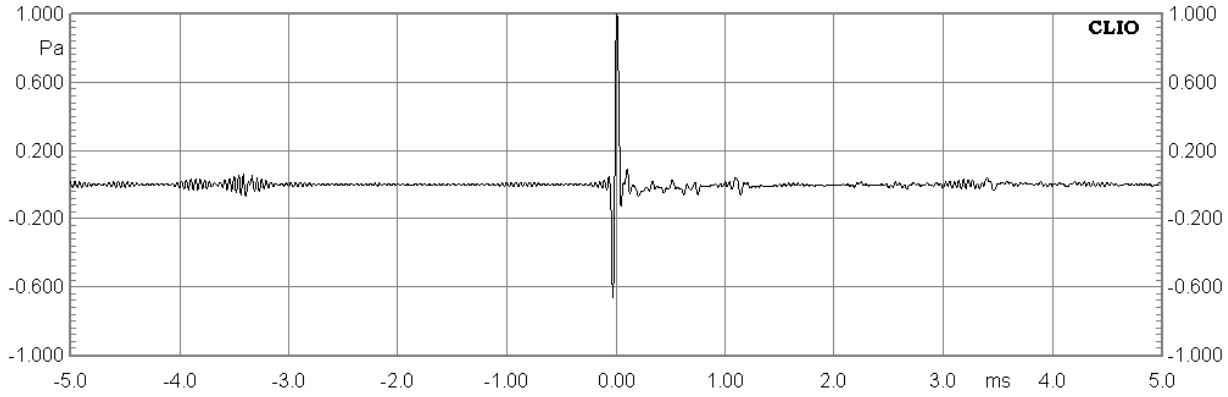
Beyond the magnitude frequency response it is also possible to measure the **phase** response and the **impulse** response. When taking acoustical measurements these functions heavily depend on the **interchannel delay** i.e. the total amount of delay present between the two channels, normally due to electronic equipment, misalignment of sound sources or flight time from speakers to microphone.

When the measurement is just started, if you select the Time data display, you may see the following impulse response.

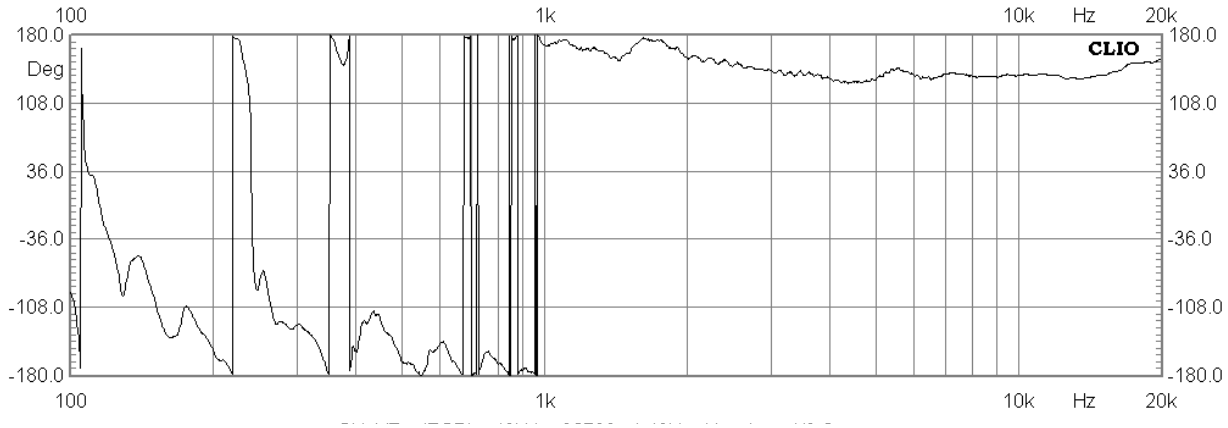


The interchannel delay is of about 3.4ms; if you want to measure the phase response of the system properly you should try to remove it. You can do this in several manners:
1) **Press the Capture delay button**; the software should calculate and automatically remove the delay setting it; you should see the captured delay in the delay display.
2) Try to input the delay value manually; you can do this **clicking on the delay display** and entering the desired value in the resulting dialog.
3) Modify the delay interactively with the keyboard using the **PgUp**, **PgDn** (0.1ms steps) or **Shift-PgUp** and **Shift-PgDn** (1 sample steps) keystrokes.

At the end of this process the impulse will be located around zero time.

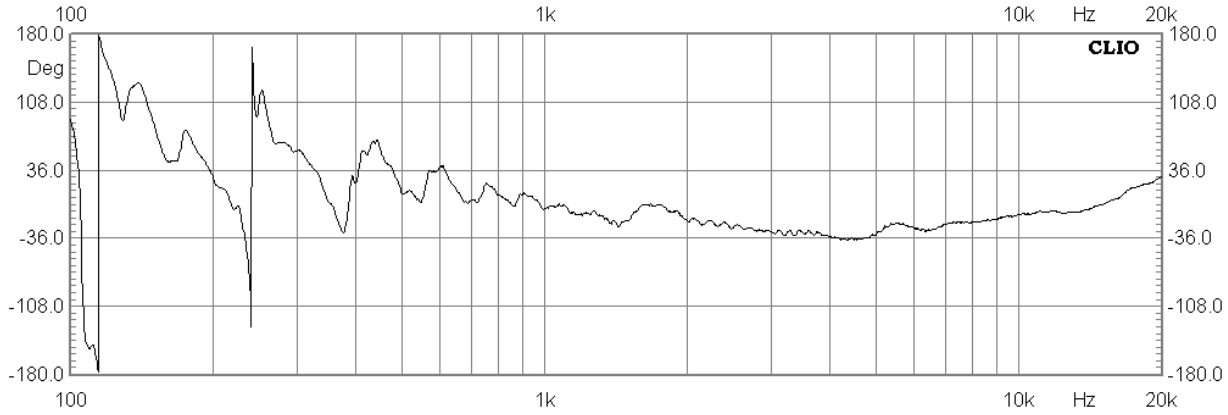


The last obstacle you may find while measuring phase is that, even if the interchannel delay has been correctly removed, still remains a phase inversion in the chain giving the following response.



It is possible to control a phase inversion with the dedicated buttons on CLIOwin desktop; simply invert the phase of either channel A or B, obviously not both!

In this way the final measurement of phase response will be as follow.



10 MLS & LOG CHIRP

10.1 INTRODUCTION

Within this menu two different technique are available that yields to the final result, the complex transfer function of a generic device. They are MLS and LOG CHIRP Analysis. While the internal processing is quite different the result is the same and this justify keeping them together. Advantages of each approach will be described later in this chapter briefly, leaving to the bibliography for details.

MLS stands for Maximum Length Sequences, is a powerful well established technique that allows you to carry out analysis of linear systems recovering the **Impulse Response** of the device using a fast cross-correlation algorithm . It is therefore a Time based analysis. Frequency domain information is obtained calculating the Fast Fourier Transform, hereafter FFT, of the impulse response.

LOG CHIRP analysis uses a log-swept sine chirp as stimuli. The Frequency Response is obtained with a deconvolution process and the impulse response with an Inverse FFT of the Frequency Response

Both approaches carry a crucial piece of information in the time domain, the **Impulse Response.** MLS & LOG CHIRP is therefore particularly well suited for recovering the anechoic sound pressure response of a loudspeaker. i.e. the frequency response of a loudspeaker as if it where positioned in an anechoic room, while carrying out the measurement in a normal room. Just as importantly MLS & LOG CHIRP allows complete evaluation of room acoustic parameters.

Within this Menu the user will be able to switch from time domain to frequency domain and back using the powerful post processing tools CLIO provides. This allows the collection of very sophisticated and complete information of any electro-acoustic device. Both the theory behind all of this and the amount of parameters that affect the measurement results make this Menu probably the most complicate to use. We will skip the theory completely and after a concise description of the whole User interface we will deal with real life applications.

10.2 MLS & LOG CHIRP CONTROL PANEL

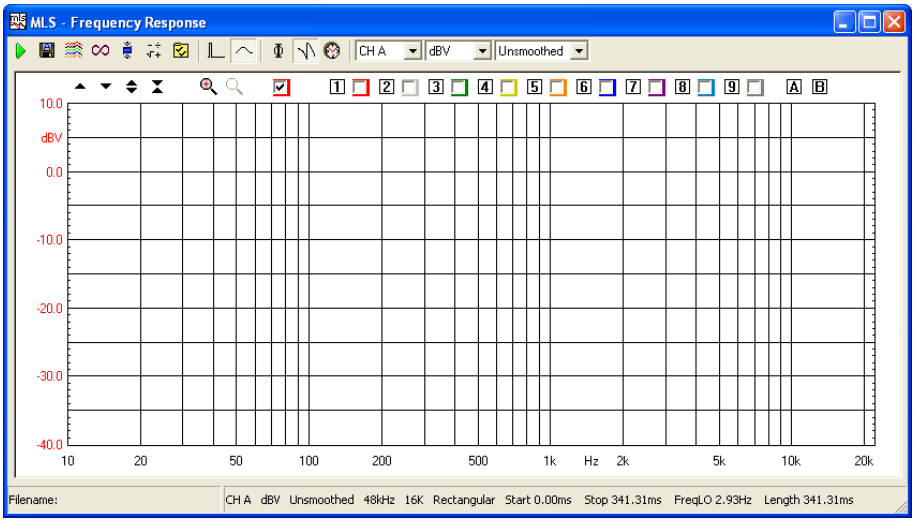


Figure 10.1

10.2.1 TOOLBAR BUTTONS

Starts an MLS & LOG CHIRP measurement.

If pressed the measurements will be autosaved. The current autosave definitions apply; see 6.3.1 for details.

If pressed the measurements will be autostored in overlays.

Selects the Loop mode. When in Loop mode the MLS & LOG CHIRP measurement is automatically repeated until the user presses a keystroke or releases the button. If Autosave is active the loop mode ends after the total files to be autosaved are done.

When an MLS & LOG CHIRP measurement is taken, it automatically applies the selected post-process.

Enters the MLS & LOG CHIRP Process dialog box.

Enters the MLS & LOG CHIRP settings dialog box.

Enters Time domain.

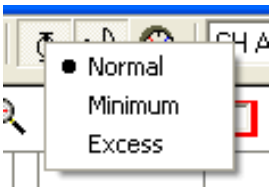
Enters Frequency domain.

Displays phase.

Set wrapped or unwrapped phase

Displays group delay.

By right clicking either on phase or group delay button the kind of calculation can be selected:



Normal displays the measured phase (group delay) curve referring to the selected time domain data.

Minimum calculates and displays the phase (group delay) curve, related to the current modulus curve, in the assumption of minimum phase behaviour (i.e. the Hilbert transform of the log magnitude).

Excess calculates and displays the phase (group delay) curve as the difference between the Normal and the Minimum ones.

10.2.2 TOOLBAR DROP DOWN LISTS

input channel
Selects the input channel configuration.

Y scale unit

Selects the measurement Y scale unit. Possible choices are dBV, dBu, dBRel as Voltage, dBSPL, dBPa, dBPa/V as pressure, dBmeter as displacement, dBm/s as velocity, dBm/s^2 as acceleration and Ohm as impedance unit. Refer to CLIO Options>Units Conversion dialog for reference sensitivities.

smoothing

Activates a frequency smoothing of the active curve. This smoothing effect will allow a better appreciation of the general features of the response curve. The smoothing algorithm that is employed averages all values within a fraction-of-octave band surrounding each analysis frequency.

10.2.3 MLS & LOG CHIRP SETTINGS DIALOG

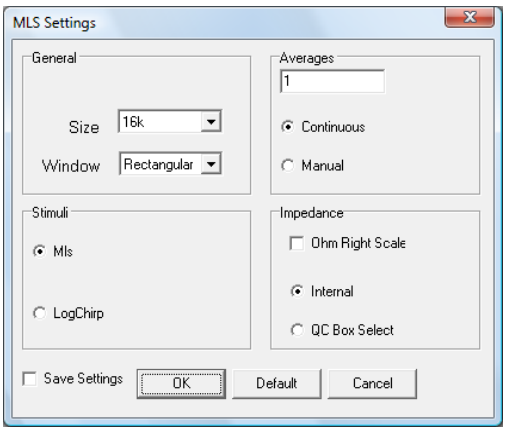


Figure 10.2

size

Selects the size of the MLS & LOG CHIRP sequence.

window

Selects the appropriate kind of window for analyzing time data. It is possible to select between a rectangular, Hanning or Blackman window; the last two can be **full** or **half size.** NOTE: These windows are applied to the time portion to be transformed with FFT. If the start point is near the impulse, **full windows** will null the most important part of the time response (due to their rise time). To evaluate the effects of a data window refer to Chapter 9, and FFT measurements in general.

stimuli

Selects the kind of stimulus, either MLS and LOG CHIRP used for the measurement.

averages

Controls the averaging mode of operation. The measurement will be repeated and averaged the number of times set, therefore obtaining a better signal-to-noise ratio at the expense of reduced measurement speed. **Continuous** performs the number of averages in the shortest time without waiting. **Manual** waits the user to press any key between each measure; it is useful, for example, in averaging different microphone positions.

impedance

Set how Impedance is calculated and displayed. When taking impedance

measurements refer either to the **Internal** impedance mode or to **QC Box Select** (the hardware setting of the QC Box determines directly the Impedance Mode, refer to 4.6). When checking **Ohm Right Scale** the impedance is referred to the right Y scale

10.2.4 MLS & LOG CHIRP POST-PROCESSING TOOLS

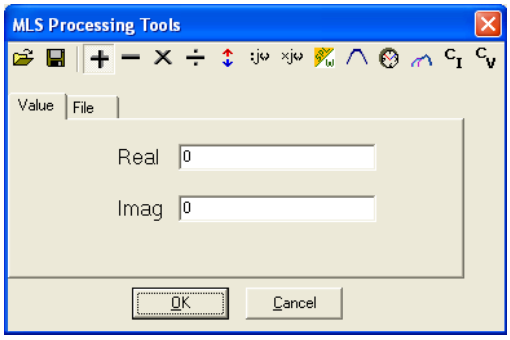


Figure 10.3

Loads an MLS & LOG CHIRP process.

Saves an MLS & LOG CHIRP process.

Adds a data value or compatible file to the current measurement.

Subtracts a data value or compatible file to the current measurement.

Multiplies the current measurement by a data value or compatible file.

Divides the current measurement by a data value or compatible file.

Shifts the current measurement by a dB value.

Multiplies the current measurement by complex frequency.

Divides the current measurement by complex frequency.

Uses a reference measurement file taken at speaker terminals to calculate 1m sensitivity in dBSPL/W. The reference file should have dBV Y units while the one in memory should be in dBSPL.

Process the current measurement with an octave band filter. It is possible to input the mid-band value and the filter bandwidth.

Temporally shifts the current measurement by a ms value. Affects phase response.

Merges the current measurement with the part below the selected transition frequency of a selected compatible file.

C_I Combines the actual measurement and the selected file to obtain a constant current impedance measurement. Both files should be in dBV.

C_V Combines the actual measurement and the selected file to obtain a constant voltage impedance measurement. Both files should be in dBV.

10.3 IMPULSE RESPONSE CONTROL PANEL

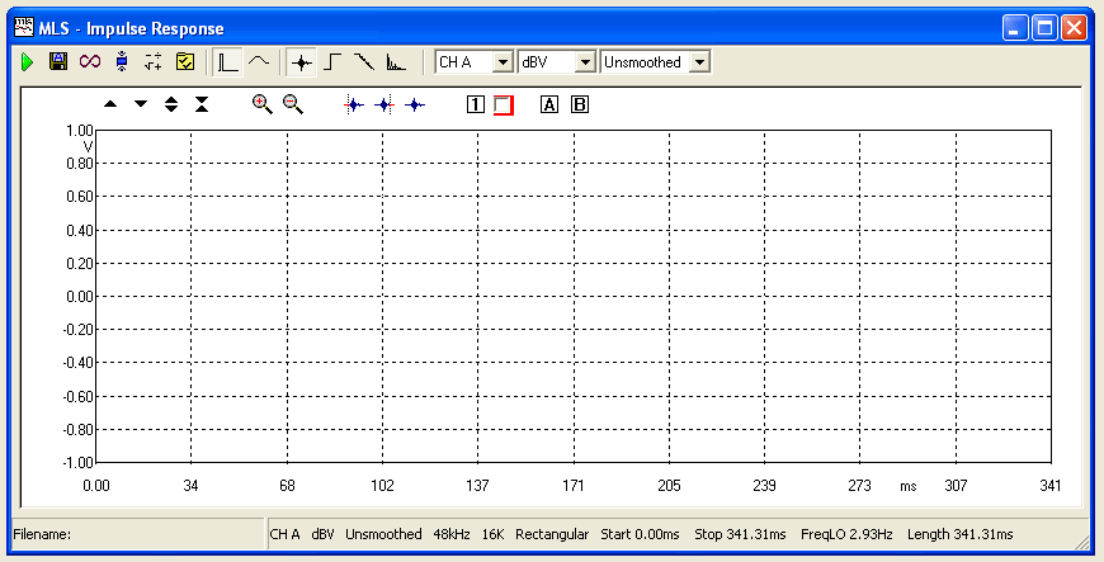


Figure 10.4

10.3.1 TOOLBAR BUTTONS

The following toolbar buttons differ from frequency domain control panel:

Displays Impulse Response.

Displays Step Response.

Displays Schroeder Decay.

Displays Energy Time Curve (ETC).

Also the following buttons inside the measurement area are particular to this control panel. See Chapter 6 for other general information.

Selects the starting point of the measurement window.

Selects the end point of the measurement window.

Restores the default state of the measurement window thus selecting all the acquired points for analysis.

1 Stores and display an overlay curve

A B Activates marker A and B

10.4 MEASURING FREQUENCY RESPONSE

In a step by step process we will deal with any single aspect that affects MLS & LOG CHIRP measurement results. At first we deal with electrical measurements, leaving acoustical as the last steps.

10.4.1 MEASUREMENT LEVEL

Opening the MLS & LOG CHIRP menu for the first time you will see a graph which has frequency on its X-axis. Our first step will be measuring the response of an "A" weighting filter. All the settings are left in their default state; we will take care when measuring level only. Please ensure that the device being measured cannot be damaged by the output level chosen for the actual measurement. In this example, the device under test cannot be damaged with an output set to 0dB (-5.21dBV with MLS signal, -2.21dBV with LOG CHIRP). Having ensured a safe level we connect CLIO output A with the device input, CLIO Input A with the device output, then enable input auto-range and click the Go button. As a result we get Fig.10.5

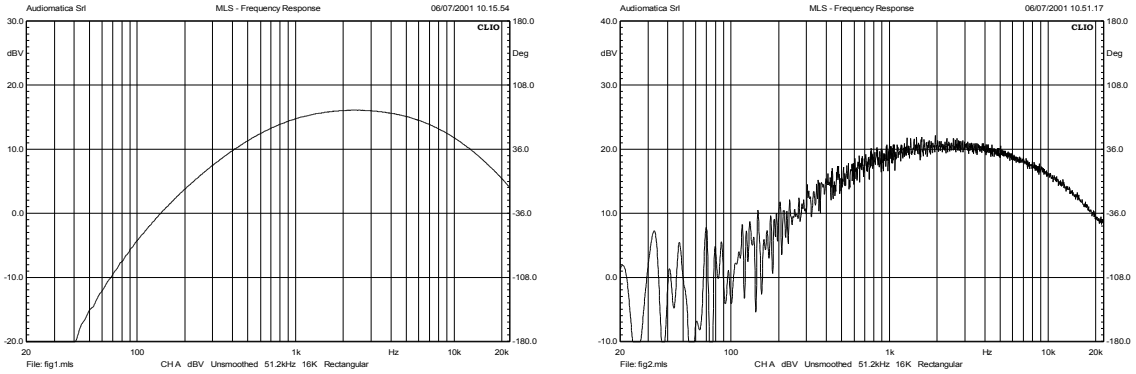


Figure 10.5 and 10.6

The curve reaches 16dBV (6.3V) at 2.5kHz, which is quite a high level for our device. Looking for trouble, we increase CLIO output to +6dB and measure again obtaining Fig.10.6. The device went into saturation; in more emphatic terms it is not linear any more. The whole MLS & LOG CHIRP process works on the assumption the device is linear. If this is not the case, it is not easy for an inexperienced user to understand what is going on just by inspecting the results. Checking the measuring level is important, especially when the gain of the device under test is unknown. You should use the Multi-Meter and the MLS & LOG CHIRP signal to accomplish this.

10.4.2 MLS & LOG CHIRP SIZE

The MLS & LOG CHIRP Size is the number of points that defines the MLS & LOG CHIRP sequence. In terms of generating a digital signal these are the number of samples before the signals is started again from the beginning. CLIO handles MLS & LOG CHIRP sizes from 4k to 256k. These terms are inherited by the computer world and are somewhat imprecise. The real size is the nearest power of 2, the 4k being 4096 points and the 256k 262144 points long (one less! for MLS, exactly this for LOG CHIRP). The first important consequence of setting the MLS & LOG CHIRP size is the length of the Impulse Response recovered which is always as long as the MLS & LOG CHIRP itself. From the users point of view what matters is how long this impulse is in terms of seconds. This in turns also depends on the Sampling Frequency set. It is easily calculated as the size divided by the sampling frequency. The default is 16k and 48000 Hz, that is 16384/48000=0.341 seconds. We will deal with this in more detail later. For now it is enough to say that CLIO also uses this

size for the FFT. This is important as the frequency resolution you get is calculated as the sampling frequency divided by the FFT size. Again for the default settings this is 48000/16384=2.929 Hz. This is already a high resolution. However thinking in terms of octave or fractions of an octave, which are the terms of a logarithmic frequency axis, 2.929Hz is around 1/2218 of an octave at 10kHz while is around 1/3 of an octave at 10Hz.

Again an example is better than a thousand words. We will measure a twin T notch filter, probably the most demanding situation for frequency resolution, with two sizes 8k and 64k, and two center frequencies, 10k and 63Hz. Results are in Fig.10.7.

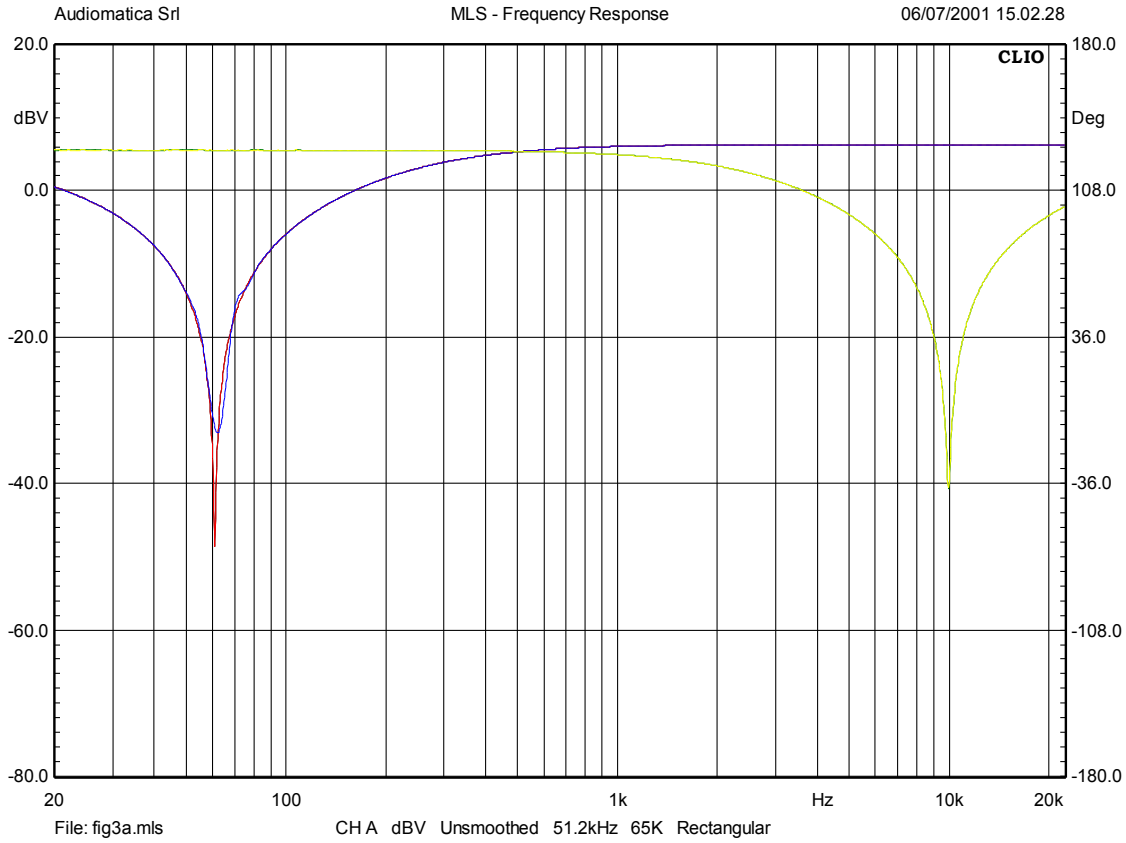


Figure 10.7

Interestingly, while we took four measurements we can only effectively see three curves. In actual fact the 8k size gives by far enough resolution at 10kHz and the associated curve is completely hidden by the 64k one. The 64k curve (Red) only differs from the 8k curve (blue) at 63Hz.

10.4.3 ACOUSTIC FREQUENCY RESPONSE

Up till now we measured using CLIO and simple cables. Now we are going to deal with acoustic measurements. The time domain will be an essential part of our interest. Furthermore we need to add two external devices, a microphone and a power amplifier. Connections are shown in Fig.10.10.

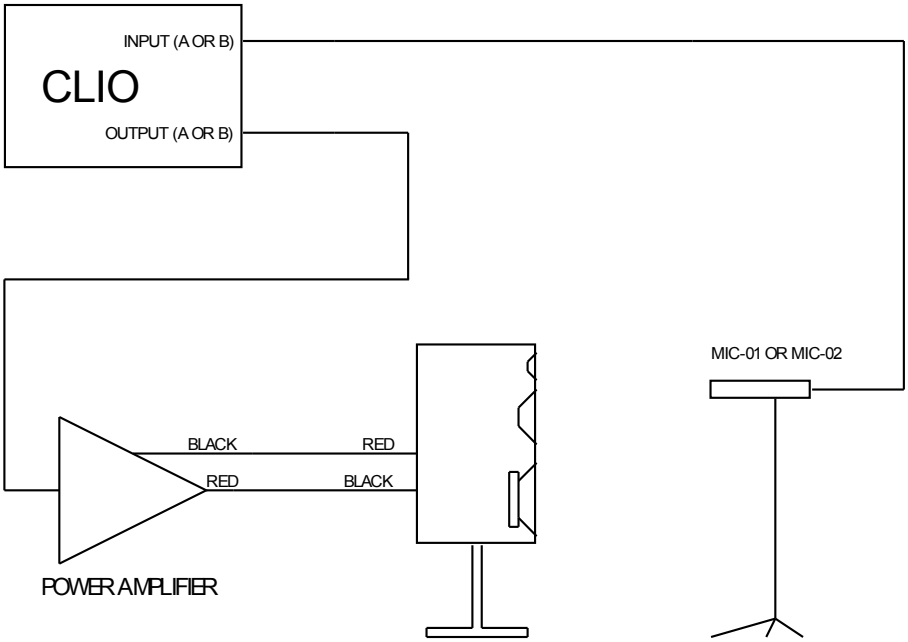


Figure 10.10

Please note that the connections from the amplifier to the speaker are inverting polarity. This is on the assumption that the amplifier is non-inverting, most are so, and you are using Audiomatica Mic-01 or Mic-02, which does invert polarity. Most Measuring Grade Microphones also invert polarity. Remember that inverting polarity does not change the frequency response, however it does change phase response. Fig.10.11 shows how we placed the microphone with respect to the speaker and also with respect to the floor.

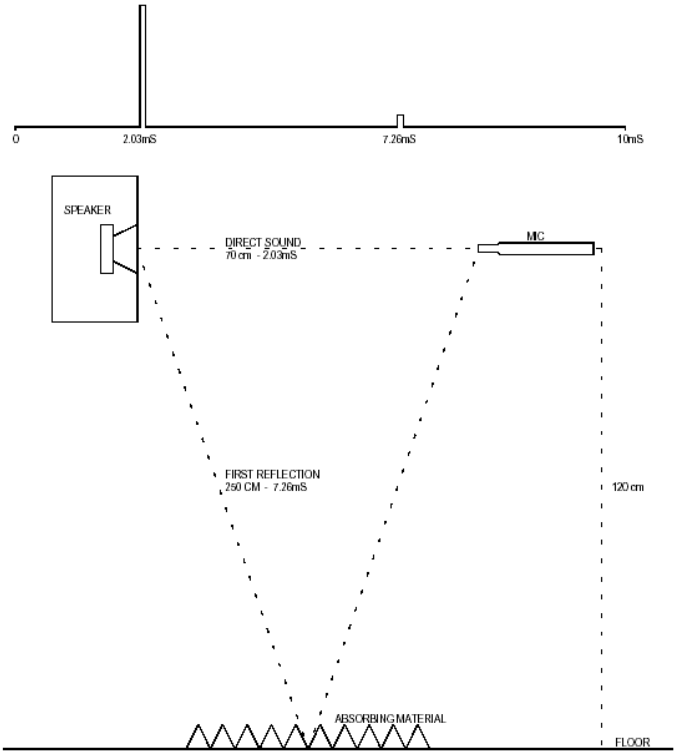


Figure 10.11

Any other reflecting surface is further than the floor. If the microphone is directly connected to the CLIO board **remember to switch the microphone power supply on.** It is also **very important** to remember to type in the correct microphone sensitivity in the microphone Dialog Box, this is crucial for setting the correct measurement level. We have already dealt with level before, however here, things are more dangerous. Supposing the amplifier used has a gain of 30dB, a common value, leaving the CLIO output level at 0dB will cause the amplifier to deliver the equivalent of circa 40W/8Ohm continuous. If the amplifier handles this power you will almost certainly burn your tweeter. If the amplifier is of lower power it is likely that it will saturate and burn your tweeter even faster! To avoid this embarrassing event, do as follow: enable CLIO input auto-range, open the Multi-Meter and select Pressure from the Parameter Drop Down. You will get the reading of your environmental noise. Fig.10.12 shows ours, we hope yours is lower!

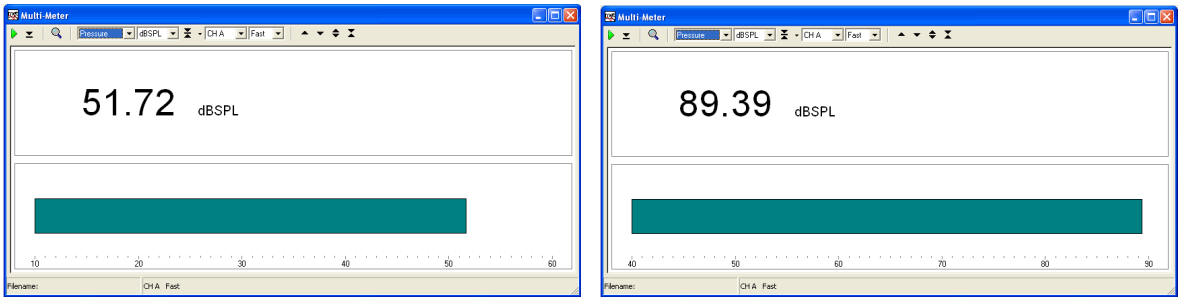


Figure 10.12 and 10.13

Now set CLIO output level to a very low value, -30 to -40dB and with the microphone positioned 70cm to 1m from the speaker start the MLS & LOG CHIRP signal. Increase the CLIO output level until you read 85 to 90 dBSPL as in Fig.10.13. Now go back to the MLS & LOG CHIRP Menu, select dBSPL as the Y scale unit and finally click on Go. The speaker we are testing is a glorious Rogers

LS3/5A, year 1978. Fig.10.14 shows our result.

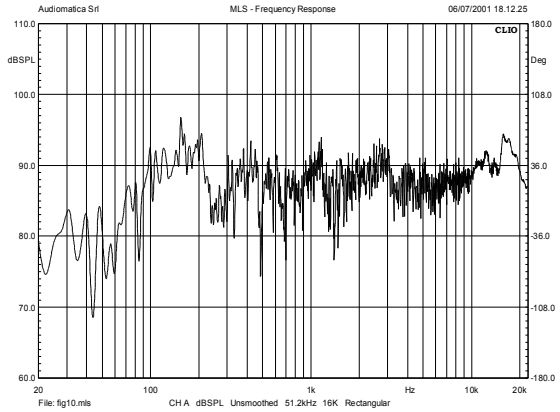


Figure 10.14

What you see is the speaker plus the room where we took our measurement, which is far from being anechoic. It is time to inspect the time domain. Clicking on the Time Domain button we get Fig.10.15 and, Fig.10.16, once we zoomed to the first 11ms and expanded the y scale.

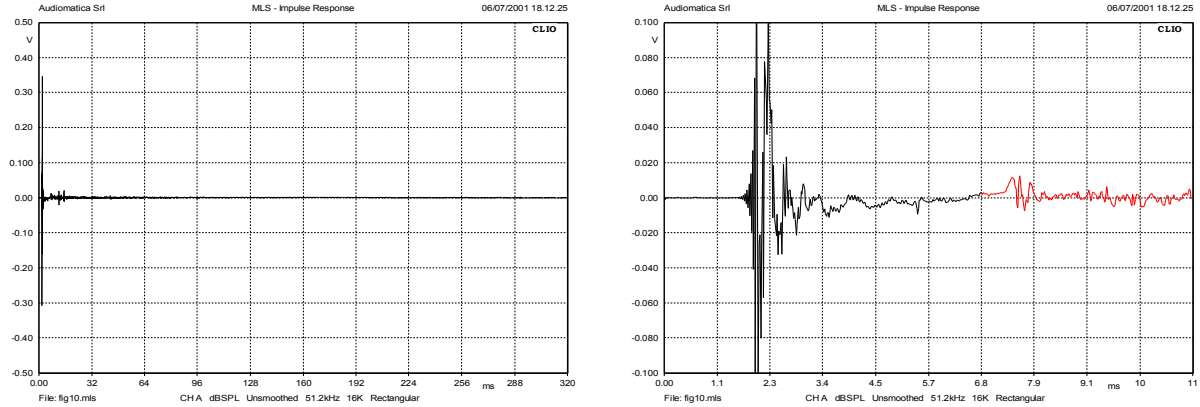


Figure 10.15 and 10.16

We also did another very important thing. At 7.3ms the first reflection due to the floor can be seen, just as expected from Fig.10.11. We set the marker at 6.8ms, which is just before the first reflection, and set the Stop Window there (see Chapter 6.6). By doing this CLIO will set all values of the impulse response received after 6.8ms to 0 before FFT is executed. In this way we simulate a reflection free environment. Clicking on the Frequency Domain Button we obtain Fig.10.17.

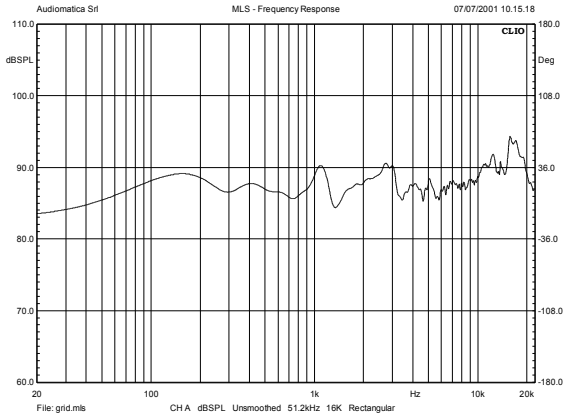


Figure 10.17

Now things look much better and this is almost the anechoic response of the speaker. However nothing comes for free. The low frequency part of the response seems quite optimistic for such a little speaker. The price we paid in setting the impulse tail to 0 is that we lost information on the lower part of the spectrum. The transition frequency between meaningful and meaningless data is calculated as 1 divided by the selected impulse length. In our case we selected a 6.8ms long impulse. 1/0.0068=147Hz right? Wrong. We have to remember the first 2 ms of the impulse, which is the time the sound takes to reach the microphone and hence does not carry any information. We could have selected the impulse as in Fig.10.18 without affecting the frequency response at all however phase response would have been greatly affected.

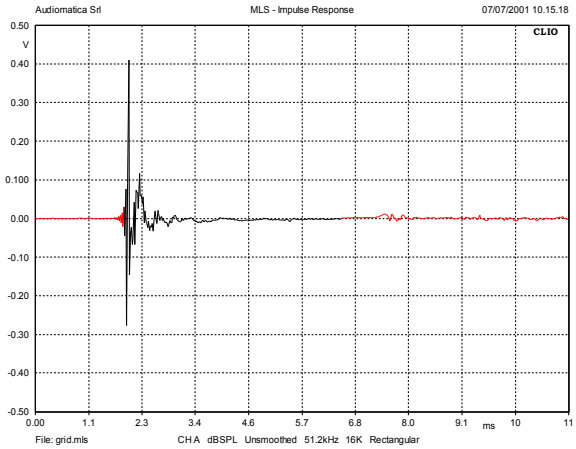


Figure 10.18

The right calculation is 1/(0.0068-0.002)=208.33Hz. In our room the smallest dimension is floor to ceiling. This is indeed the most frequent case. This dimension is however 4m. The best location for the speaker would have been at 2m both from the floor and the ceiling. The second consideration is microphone distance. The further away it is, the more you have to subtract from the impulse length due to sound travel time to the microphone. In practice we do not encourage distance below 70cm for complete speaker measurement and you should increase to 1m for bigger ones. However single driver measurement can take advantage from a reduced distance.

10.4.4 PHASE & GROUP DELAY

We used the term "Frequency Response" to refer to graphs of Fig.10.5 and Fig.10.8. Frequency is in the x-axis in both figures. The units that respond to frequency, y-axis, are Volt and Ohm, respectively. Both of them are complex quantities (have real and imaginary parts) and their **magnitude** is shown. Doing this we obtained a very useful piece of information but we lost the original data (infinite numbers of different real and imaginary part can lead to the same magnitude). How this information loss will affect your results depends on what you are going to do with these graphs, or, better still, what the original question you were trying to answer was. Referring to Fig.10.5. A reasonable question could have been: how much does "A" filter attenuate a signal at 100Hz in respect to 1kHz? You go through the graph with the marker and answer 19.3dB. If you have the IEC 651 norm you can go to the "A" filter specs and you will find this attenuation should be 19.1dB +- 0.5dB for type 0 devices. End of your job. Let's now pose another question referring to Fig.10.8: by how much would a 10kHz tone would be attenuated if we wire a 10 ohm resistor in series with that woofer? We are simply not able to give the correct answer. We need another piece of information, which is **PHASE**. Fig.10.19 is the same measure of Fig.10.8 with phase curve overlaid. To obtain it we just stored the magnitude curve and clicked on the phase button.

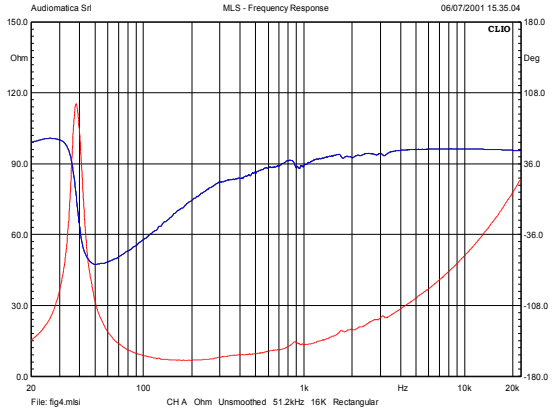


Figure 10.19

The same principal applies to acoustic devices. Fig.10.20 shows the magnitude response of a woofer and tweeter in a box without a cross-over network.

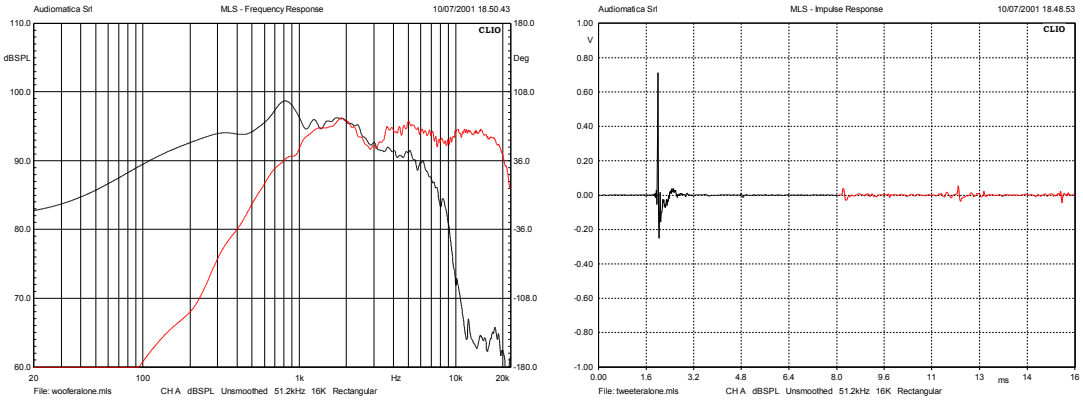


Figure 10.20 and 10.21

The two measurements were taken from the same microphone position. If we were to we ask ourselves which is the summed output we could not answer from the magnitude data alone. Unfortunately acoustic phase is not so easy to handle as electric phase. We are going to base our example on the tweeter, whose impulse response and window settings are in Fig.10.21. The procedure for the woofer would

be exactly the same.We will take this opportunity to introduce the use of the Wrapped Phase Button. Figures 10.22 and 10.23 shows the tweeter phase curve, unwrapped and wrapped.

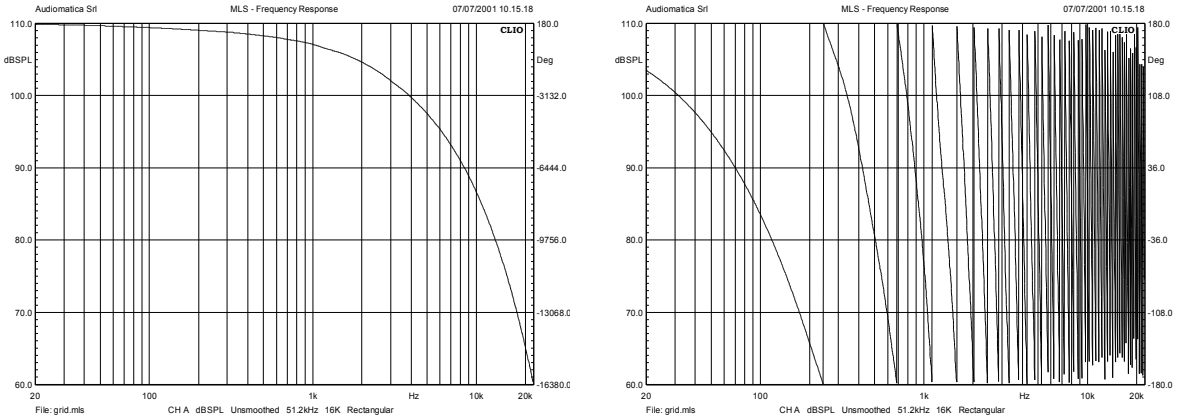


Figure 10.22 and 10.23

The reader may wonder if these figures are correct at all and if they have the same usefulness, at least visually. Well, the curves are absolutely correct; their visual usefulness is zero for the wrapped curve and low for the unwrapped. Difficulties in getting simple visual information from these curves arise because they are the sum of two effects. The first one is the devices own phase response. The second is the time of sound flight. The latter does affect the curves much more than the first one, completely burying it. The good news is that it is often possible to separate these two effects. However, the bad news is that this is not an easy task. Trying to explain it, without going into heavy mathematics is very difficult. The bibliography in this user manual should be considered as an integral part of it here. Within CLIO the time of flight can be removed in several different ways, with different degrees of accuracy. The most accurate is also the most complicated and is how we are going to proceed. Fig. 10.24 introduces us to **Minimum Phase**, which is the heart of the whole procedure.

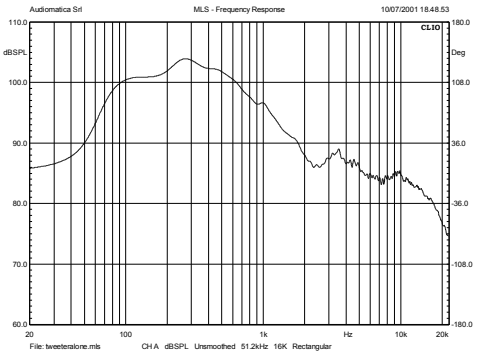
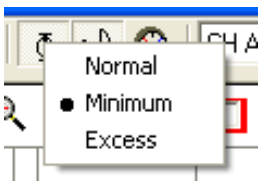


Figure 10.24

We obtained it by selecting minimum phase in the MLS phase Drop Down Menu (right click on the phase speed button).



Certain well-behaved systems are defined as Minimum Phase. In these, the phase

response can be obtained from the magnitude response by calculation. Another kind of phase (we promise it is the last one), is **Excess Phase**. This is the algebraic difference between true phase, as in Fig.10.22, and minimum phase. It is exactly what we need to separate the time of flight from the devices own phase response. We won't use excess phase directly here but a post process of it, **Excess Group Delay**. Fig.10.25 is the excess group delay of our tweeter vs. frequency.

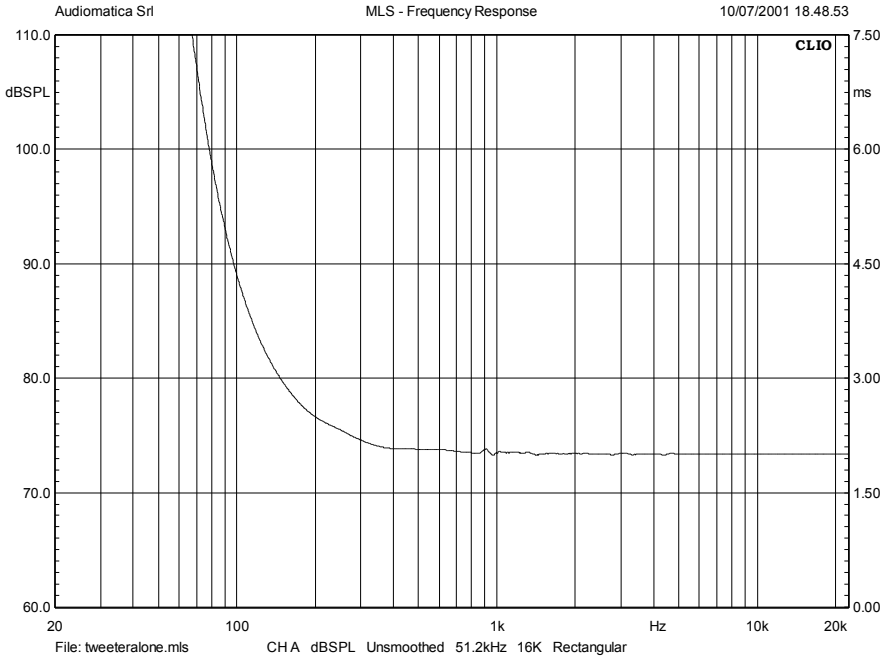


Figure 10.25

It was obtained by selecting Excess in Drop Down Menu. This graph represents the distance of the sound source from the microphone vs. frequency. As long as the distance is constant the system is minimum phase and we are in the position of a well defined acoustic center. If you recall from previous paragraphs we have reliable data down to 200Hz because of the time windows. As we deal with a tweeter we will consider the 2k-20k frequency range where the marker reads a constant 2.01ms. We will use this value to operate a time shift that removes the sound flight time. This is accomplished from the Processing Tools Dialog, selecting Time shift and typing the value we found as in Fig.10.26.

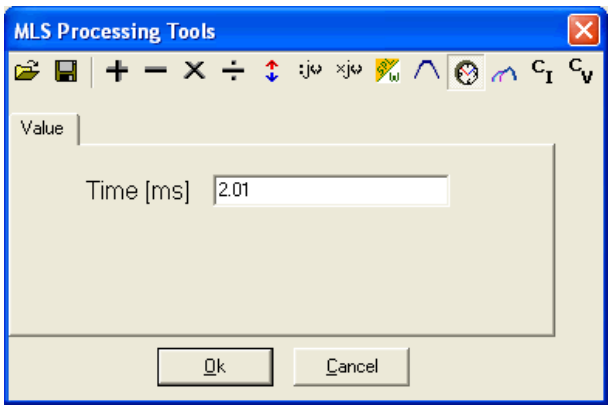


Figure 10.26

Clicking OK we can finally display the processed phase, minimum phase and magnitude response of our device in Fig.10.27. We used the term "processed phase" here and this is not casual.

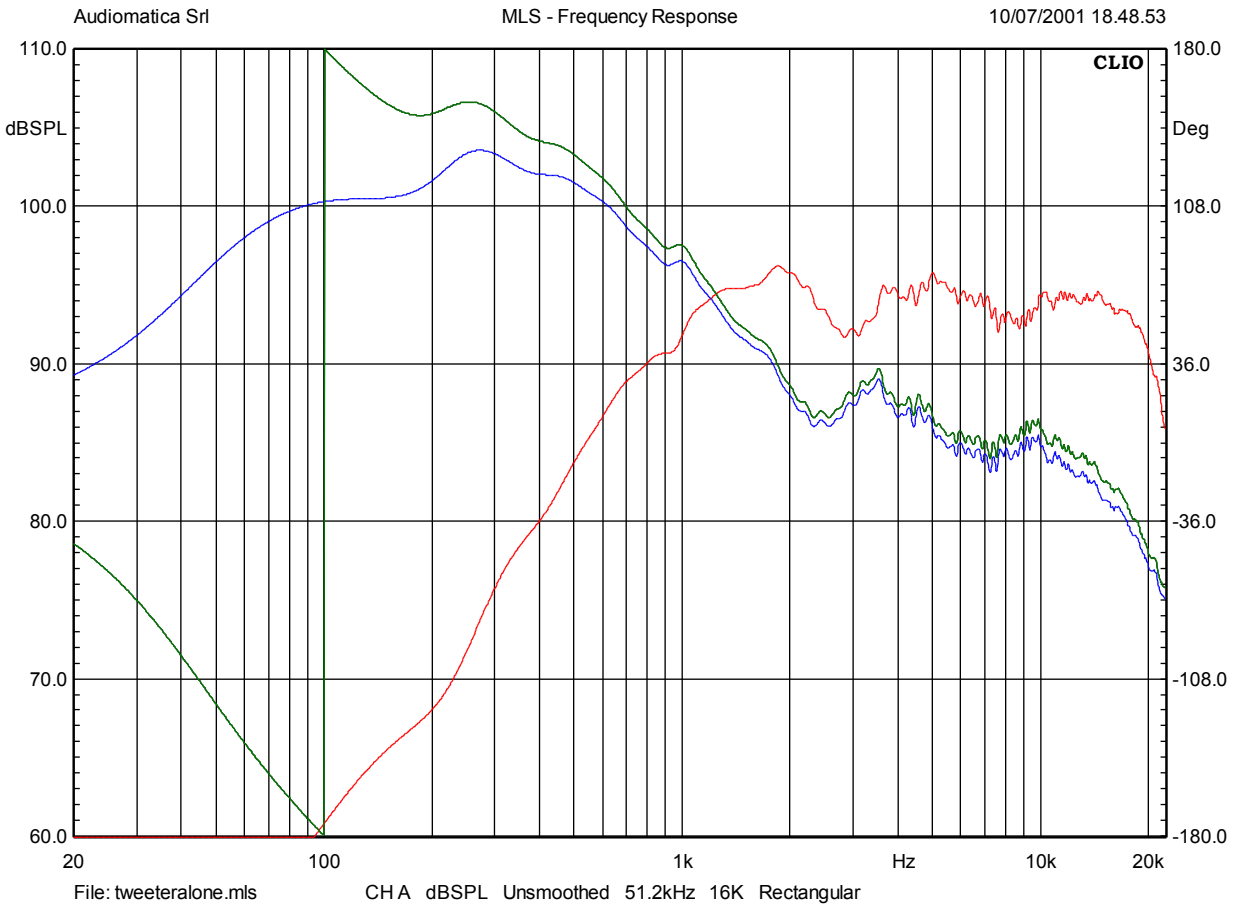


Figure 10.27

To finish this difficult paragraph we will summarize what we did with some comments. Measuring acoustic phase response is often far from a "press a button and get it" procedure. We went through several phase plots, all looking different but, this is an important point, all correct. It is common to identify the processed phase as the true one only because it looks better. It is important to stress that the true phase is that of Figures 10.18 and 10.19. CLIO, which is intended as a computer based instrument, can, as we will see later, easily calculate the summed response of woofer and tweeter after they are taken separately but with the same microphone position. What we did with our complicated procedure was to obtain a response as if the microphone would have been exactly in the acoustic center of the driver. The most obvious application is to furnish data to cross-over CAD programs.

10.5 OTHER TIME DOMAIN INFORMATION

Besides the impulse response we already dealt with, CLIO gives three more time related post processing, which are ETC, Step Response and Schroeder Plots. The last is room acoustic oriented and we will handle it later with a T60 calculation example. ETC and Step Response are shown here, Fig.10.28 and 10.29; they are relative to the system of Fig.10.15.

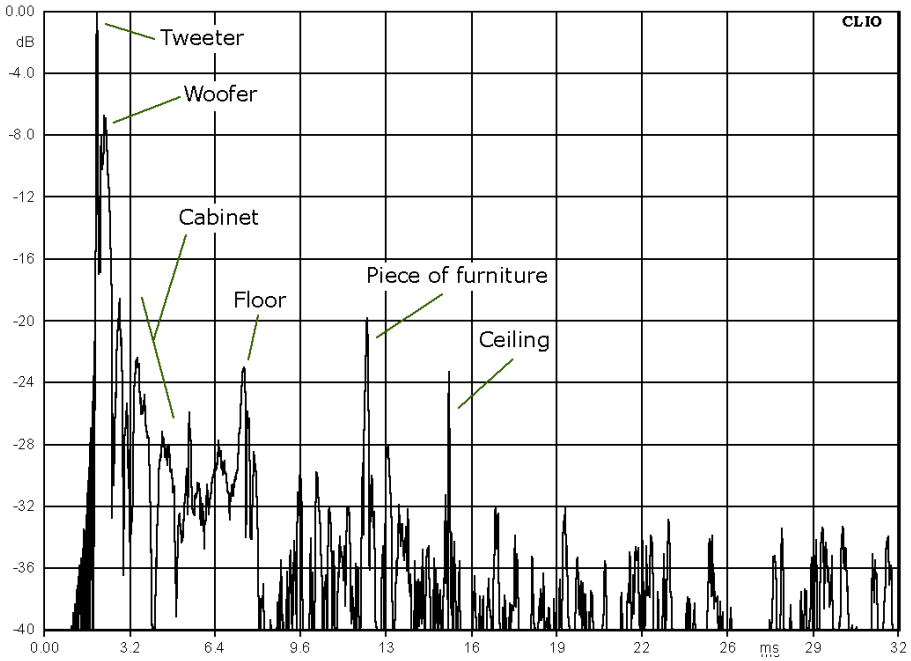


Figure 10.28

Interpretation of ETC, besides the most obvious things which are labeled inside the figure, requires some theoretic digressions beyond the scope of this user manual as the topic is still debated.

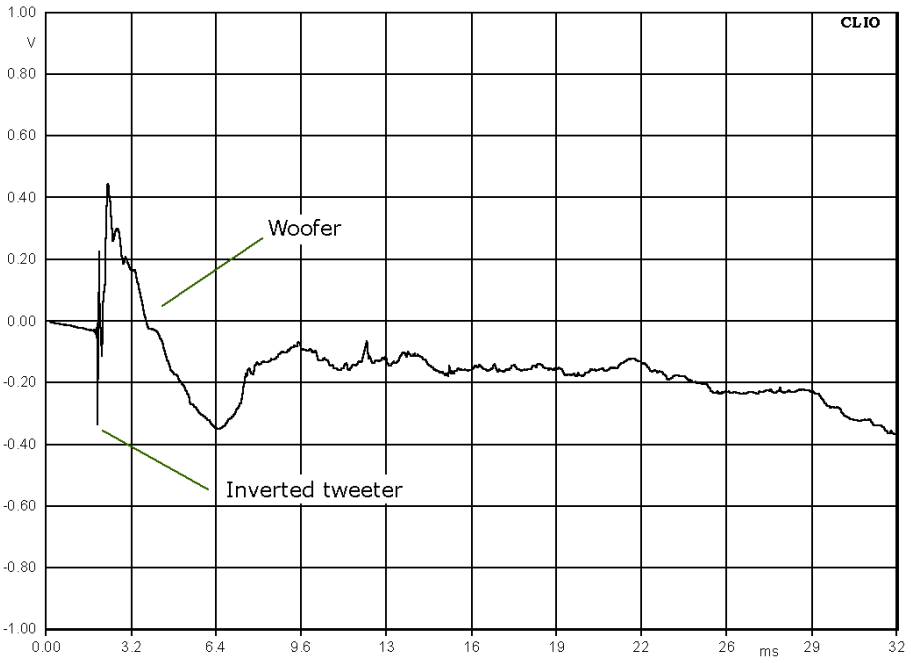


Figure 10.29

Step response is very useful to identify differences in arrival time between drivers as well as their polarity.

10.6 PROCESSING TOOLS BY EXAMPLE

CLIO has powerful processing tools that can be helpful in several circumstances. We saw the basics at the beginning of this chapter. It was just a brief description of the kind "press this to do that". Here we are going to use some of them in practice. Some general rules apply to a group of them for four basic operations. You can add, subtract, multiply and divide the data in memory either with a single complex value or with a **compatible file**. Compatible means that it must be a file with the same basic settings as the one in memory. Y-scale must also be the same that is you cannot add dBV with Ohm. The use of Load Process and Save Process, will become clearer as we review the examples. The Add and Sub functions are used mostly among files. We will use them to obtain the sum and difference of the woofer and tweeter of Fig.10.20. Results are in Fig.10.30 where the sum is in red and the difference in blue. These are useful plots to start with during a cross-over design. For example it is interesting to consider the big notch in the sum curve, something not easy to imagine from the magnitude response.

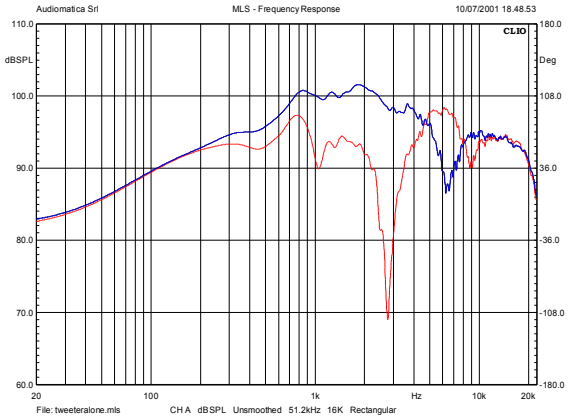


Figure 10.30

The divide function, in its most classical use, allows the user to show a magnitude response as a transfer function with another measurement. Suppose you want to evaluate how the grid affects the frequency response. We will use the response of Fig.10.17 as a reference, which has been taken with the grid in place. We remove it, take another measurement, and perform a division by file with the grid in place. We remove it, take another measurement, and perform a division by file with the data of the reference. Result is in Fig.10.31.

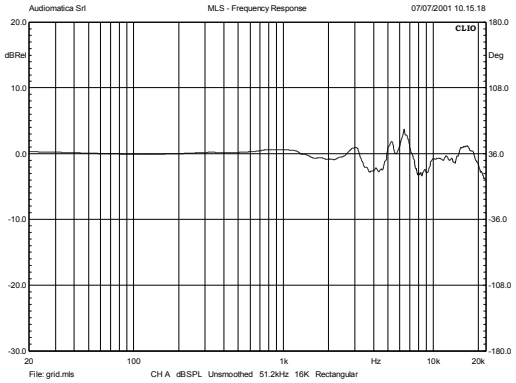


Figure 10.31

Note that the y-scale has been automatically changed to dBRel. This is a good point to introduce the use of Load and Save process together with the Automatic Processing button. Any process you execute can be saved to disk and will have

"mpro" extension. This allows you to recall any value or file path later on by loading this file again. Suppose you have a small production of ten speakers that you want to test against a previous produced reference which you know is fine. You just have to define and save a process that divides the current data with the reference. Testing a device against itself should produce a flat line, within the frequency range of the device, and this should be checked before saving the process. When you need to check your batch you just recall your process. This will activate the Automatic Process button. When this button is activated any subsequent measurement is processed before it is displayed. The next process we will consider is the **dBSPL/Watt**. It requires a file and an impedance value and allows us to obtain a frequency response plot referred to 1W at the speaker terminal whatever the real measurement level is. To make this possible an electrical measurement at the speaker terminal (power amplifier output) must be taken, with dBV as y-scale, and used as a reference file. A value of impedance is also necessary that allows CLIO to convert voltage to power. We will go through the entire procedure using the tweeter of the previous examples. Since what we are looking for is the response in a 1W-1m condition we have to move the microphone to a distance of 1m from the tweeter (it was at 69.14cm). The 1m condition can be set directly with a meter. Fig.10.32 shows the 1m measurement in black and the previous one in red. Signal level at speaker terminal is unchanged. The average difference being 3.3dB.

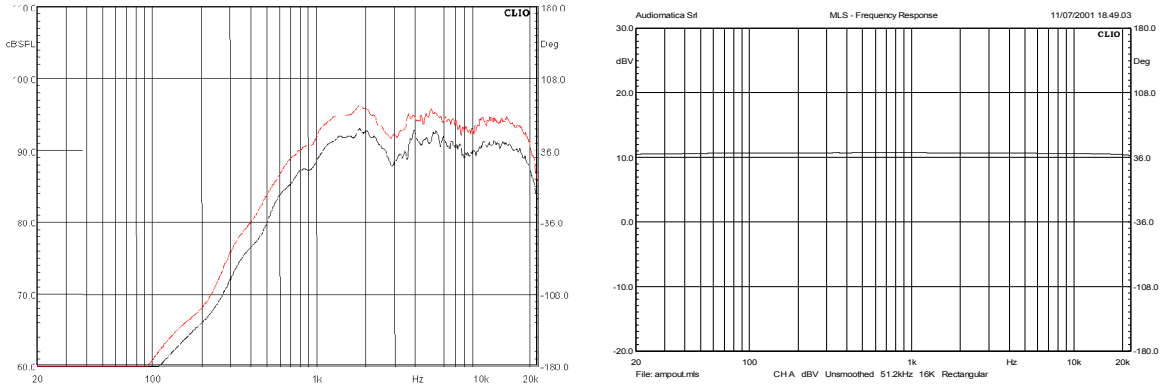


Figure 10.32 and 10.33

We now connect CLIO's input to the speaker terminals and change the y-scale to dBV, **do not change** CLIO's output level, we then obtain Fig.10.33. This is a nearly flat line that indicates the voltage delivered to the speaker. Remember to save the measurement to disk. It is worth pointing out that deviation from linearity in this curve due to the power amp or cable would be compensated for by this procedure. Now we reload the response at 1m, go to the Process Dialog that will appear as in Fig.10.34.

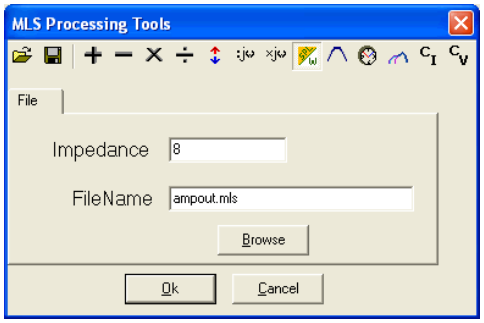


Figure 10.34

We set the impedance to 8 ohm as this is the nominal impedance of the tweeter. Clicking OK we obtain Fig.10.35, which is the final result.

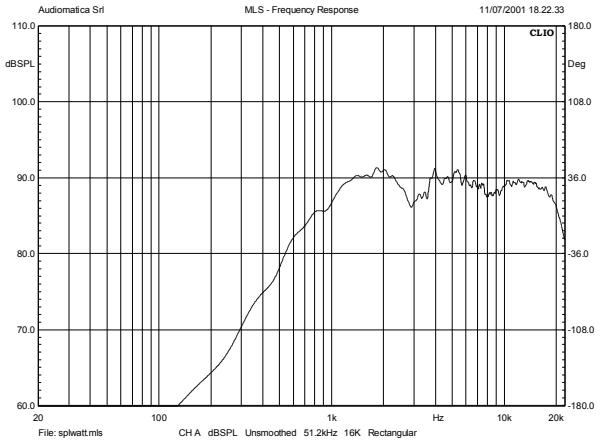
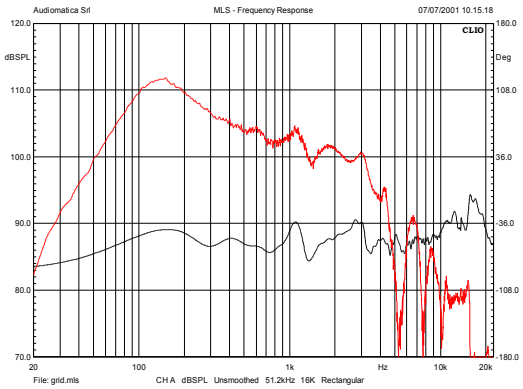


Figure 10.35

Our last example will cover the **merge function**. When we measured the system of Fig.10.17 we stated that the lower frequency limit that had to be considered reliable was 208Hz. We can easily overcome this limit taking another measurement with the near-field technique and merge the result with the data of Fig.10.17. Doing near-field measurement you should readjust the output level so that the maximum sound pressure does not exceed 110dBSPL as this will prevent Microphone overload problems. Fig.10.36 shows both the near-field and far-field response. Notice the big level difference. To merge them it is the far-field measurement which should be the one in memory. Fig.10.37 shows the Merge Dialog with a transition frequency set to 350Hz.



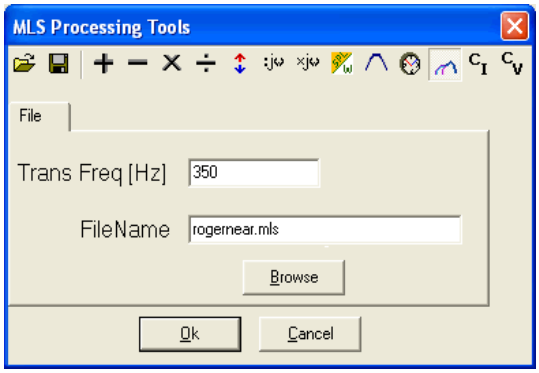


Figure 10.36 and 10.37

Fig.10.38 shows the merged response. Notice that the near-field measurement level has been scaled to the far-field one. Two very important functions, **CV function** and **CI function**, are explained in Chapter 13 as they are impedance tools.

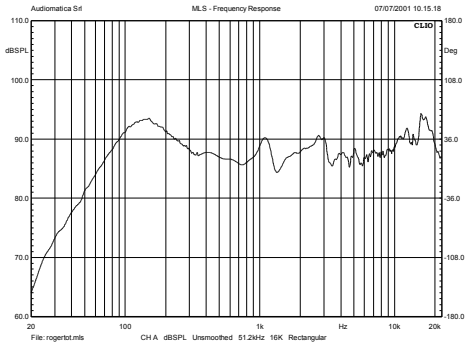


Figure 10.38

10.7 MLS Vs. LOG CHIRP

As anticipated in the introduction, some advise are given to help choosing between MLS and LOG CHIRP stimuli. Both approach are valid and bring to equivalent results. In both cases the device we want to measure is assumed to be Linear and time Invariant. This assumption while reasonably true in general cases in never met in absolute terms. There is always a certain degree of non linearity and, in less degree, a time variance. The two approach shows different sensitivity to these facts. Advantage of MLS are that is that much less computing requiring, historically much more widespread and known and has a flat frequency energy distribution. The latter feature can be an advantage or not depending on situation/DUT.

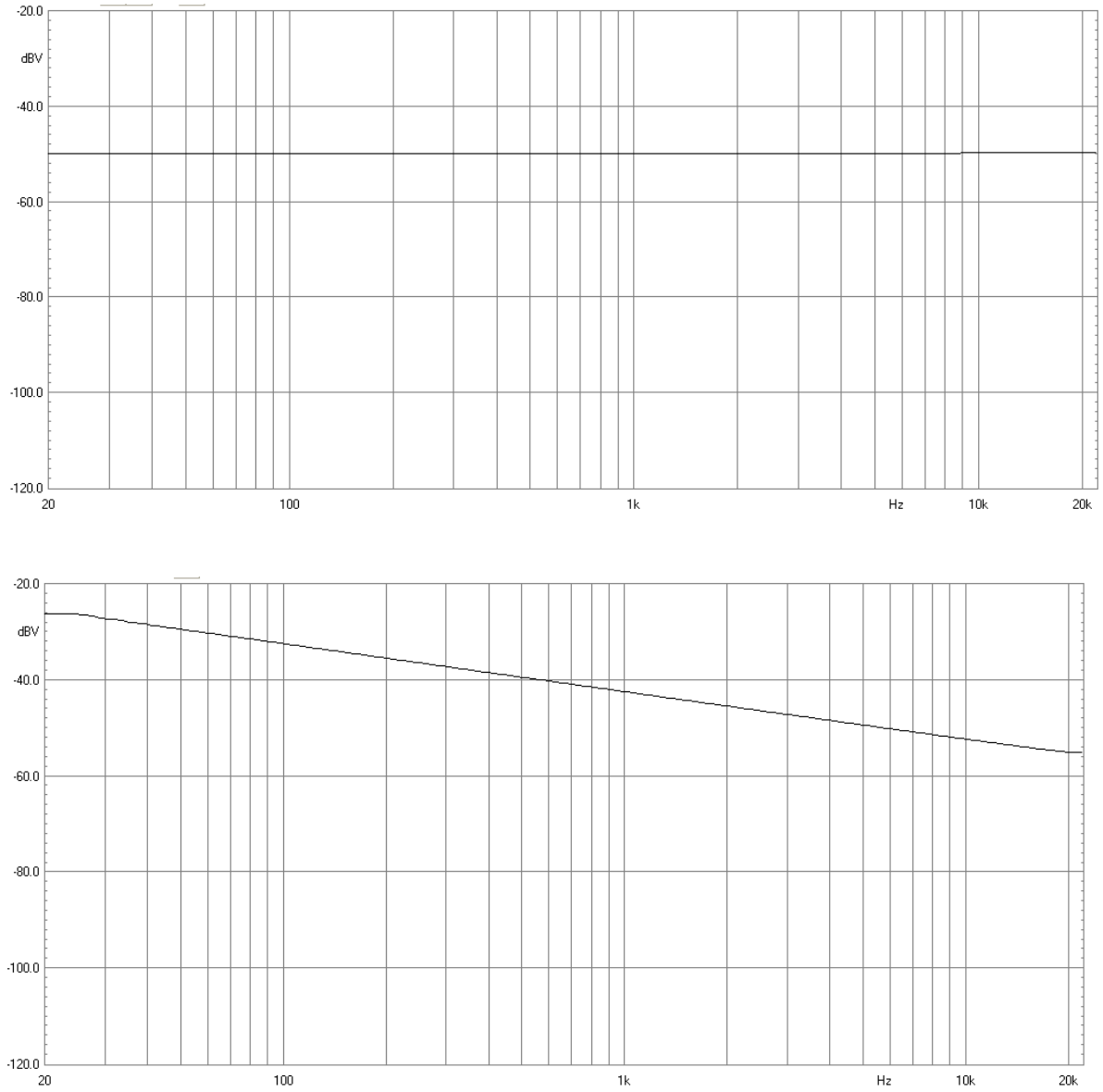


Figure 10.39

In the above figures the spectrum of both signals is showed. As usually signal to noise ratio get worse at lower frequency, the higher signal energy that LOG CHIRP has here is an advantage for S/N ratio. If the DUT does not appreciate low frequency high energy content, i.e. a tweeter, the advantage become a disadvantage.

Also, there is an important difference on how non linearity affects the two procedure. In MLS distortion transforms itself in noise spread over the whole

impulse response while with LOG CHIRP concentrate itself in single impulses (one for each harmonic) in the tail of the impulse and can easily be manually removed.

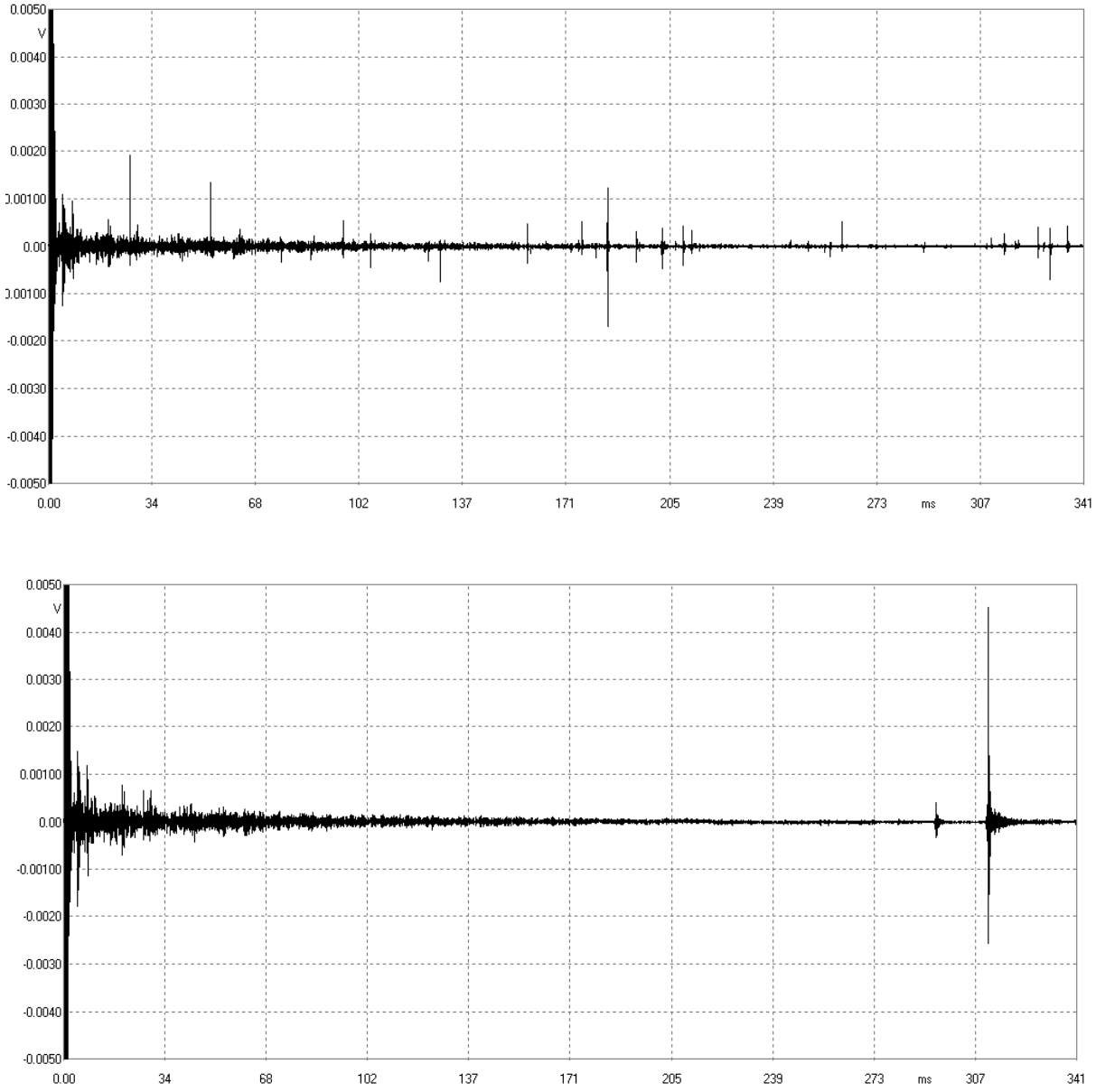


Figure 10.40

Finally a brief note on level. Setting the CLIO's output level to 0dB you'll have –5.2dBV with MLS and -2.2dBV with the LOG CHIRP, exactly the same level that you'd have within the Sinusoidal menu. While this is our choice, it is a by-product of the fact that MLS, in real life, has a higher crest factor than a sinusoidal signal.

10.8 RELATED MENUS

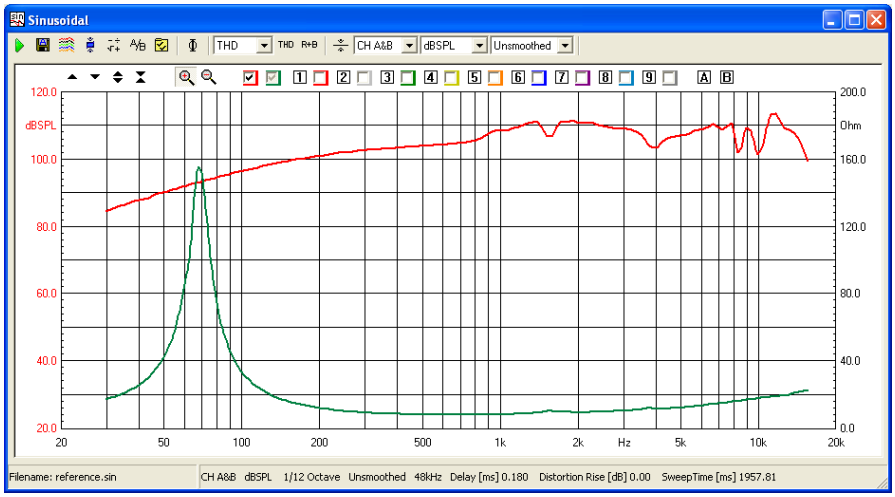
The dual domain data, Frequency and Time, obtainable within this menu, are the starting point for many kind of post processing. While some can be done within MLS & LOGCHIRP, using the **Processing Tools,** the Time Domain features (ETC, Schroeder Decay, Step response, window selection, transform start and end points) both complexity and results presentation flexibility suggested to implement other very important post processing in separate menus. These are **Waterfall&Directivity & 3D, Acoustical Parameters and Wavelet**. Each has its own dedicated chapter explaining how to use this powerful Post Processing features. However they completely rely on measurement you have done here. The ill-famed saying "rubbish in rubbish out" apply. Before dealing with post processing, be sure to have reliable measurements, with good S/N ratio, optimum size and Sampling Frequency, as pointed out during this chapter.

11 SINUSOIDAL

11.1 INTRODUCTION

Within Sinusoidal, it is possible to carry out Frequency Response Analysis, Impedance Analysis and Distortion Analysis. As should be obvious the stimuli used is a Sinusoidal signal, stepped or continuously swept within user defined Frequency limits. Although Sinusoidal steady state analysis is among the oldest and more traditional kind of measure, CLIO merges the reliability of this well known technique with the power of advanced DSP. The completely programmable Gating feature allows the user to add quasi-anechoic frequency response capability. Simultaneous, two channels, stereo measurements can be performed.

11.2 SINUSOIDAL CONTROL PANEL



11.2.1 TOOLBAR BUTTONS

Starts the sinusoidal measurement.

If pressed the measurements will be autosaved. The current autosave definitions apply; see 6.3.1 for details.

If pressed the measurements will be autostored in overlays.

Automatically applies the defined post-process after the measurement has been taken.

Enters the Sinusoidal Post-Process Dialog, described below.

When a stereo measurement is done divides channel A by channel B and shows result.

Displays the phase response instead of modulus response.

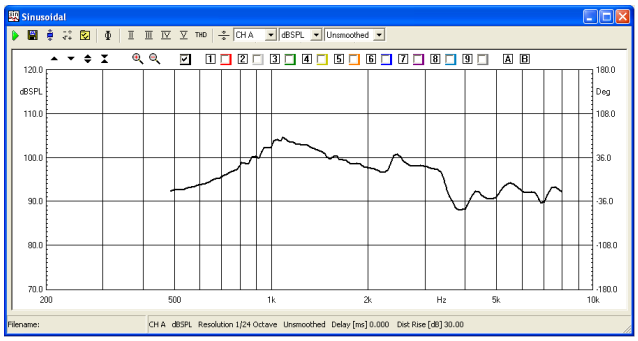
Enters the Sinusoidal Setting Dialog, described in detail below. It is the heart of the whole menu and should be thoroughly understood before pressing Go.

Harmonic Distortion Selection Drop-Down
If the distortion products have been calculated, interactively select the one to display.

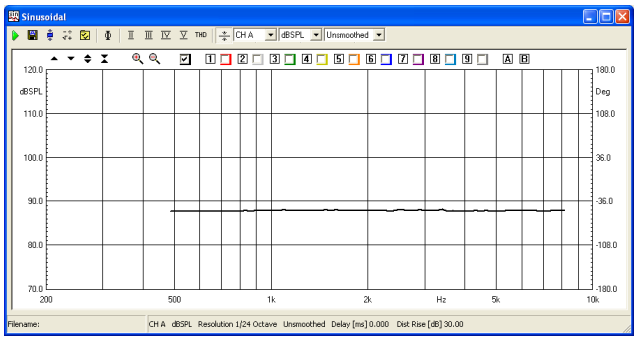
THD Displays total harmonic distortion, risen the dB defined in the Settings Dialog.

R+B Displays Fast-Track Rub&Buzz, risen the amount of dB defined in the Settings Dialog. Note: Fast-Track Rub&Buzz is available only in QC software version.

Set output level equalize mode; after a sinusoidal measurement has been taken it is possible to refer to the acquired frequency response in order to generate a colored output that flattens out the subsequent response. For example if the following pressure response has been measured:



You obtain flat output from the same loudspeaker pressing the equalize button:



11.2.2 TOOLBAR DROP DOWNS

input channel

Selects the input channel (**CHA** or **CHB**). Choose **CHA&B** for a stereo measurement when the two channels are measured and processed simultaneously.

Y Scale units

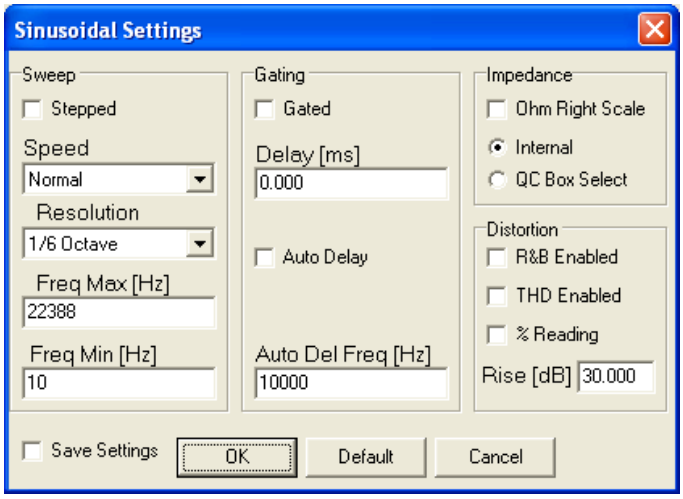
Selects the measurement Y scale unit. Possible choices are dBV, dBu, dBRel as Voltage, dBSPL, dBPa, dBPa/V as pressure, dBmeter as displacement, dBm/s as velocity, dBm/s^2 as acceleration and Ohm as impedance unit. dBV and dBu refer the 0 dB on the scale to 1 V and 0.775 V, respectively. dBrel refers to the 0 dB on the scale to the value set within the MULTI-METER. Refer to CLIO Options>Units Conversion dialog for reference sensitivities. Once the measurement is taken the conversion is done and all the internal data is stored in the relative SI unit. This means that further changes global sensitivities do not affect any data in memory or saved measurement. Ohm switches the system to convert the measurements in Ohm basing the conversion on the Impedance Mode Settings available in the Settings Dialog. When stereo measurements are taken both channels share the same unit. **Note: there is an important exception to scale management; when a stereo measurement is taken and the sinusoidal setting "Ohm Right Scale" is selected, then channel B**

measures impedance sensing current from a dedicated QCBox ISense output (Internal Mode is not allowed as DUT must be connected to amplifier's output).

Smoothing

Allows the user to select a Frequency smoothing of the active curve. The smoothing algorithm averages all the value within the selected fraction of octave band, surrounding each analysis Frequency. It is a non destructive post process that can be applied or removed at any moment after the measurement has been taken. **Note: Smoothing is not active for Ohm scale and for Rub&Buzz.**

11.2.3 SINUSOIDAL SETTINGS DIALOG



This is undoubtedly the most important User Interface of the entire Menu. Here the single action of every control is considered. Theoretical and practical effects on the subsequent measurement are considered later.

Sweep Settings

Stepped Check Box

Allows the user to choose between continuous or stepped Logarithmic Sweep. Continuous sweep is faster; the user should have clear how this may affect the measurements.

Speed Drop Down

Lets the user choose between Slow (max accuracy), Fast (optimized for Fast-Track Rub&Buzz) or Normal measurement speed. The resulting sweep time length in seconds may be read in the sinusoidal control panel status bar.

Resolution Drop Down

Lets the user choose between five different Frequency resolutions. It affects the measurement execution time, whatever the other settings are.

Freq Max Edit Box

Lets the user define the highest Frequency of the Sweep. This is also the starting measurement Frequency. The highest value accepted is 22388 Hz. The value should also be one octave higher than the Minimum Frequency.

Freq Min Edit Box

Lets the user define the lowest Frequency of the Sweep. This is also the end measurement Frequency. The lowest value accepted is 10 Hz. The value should also be one octave lower than the Maximum Frequency.

Gating (Acquisition) Settings

Gated Check Box

Lets the user enable the gating acquisition mode. Checking it will automatically check Stepped Check Box. That is, Gated Measurements are always carried out in Stepped Mode.

Delay Edit Box

Lets the user define the delay, in ms, applied between the signal generation and its acquisition. When different than 0, gating is active, even when gating or Stepped check boxes (but not both) are not checked. Typical gated use is quasi anechoic Frequency Response Analysis where it removes the time delay of the sound leaving the Speaker and reaching the Microphone. Alternative use, with gated not checked, might be removing the delay between the play and recording head in a three heads tape recorder as well as any digital processor that introduces delay in the signal path. When the settings are Not Stepped and Not Gated the delay value is reset automatically to 0; CLIO introduces a large delay anyway between start of generation and acquisition. The highest Delay value accepted is 320ms.

Auto Delay Check Box & Auto Delay Frequency Edit Box

If Auto delay is checked, when delay is used (see above) CLIO tries, using the Frequency entered in the Edit Box, to determine the delay automatically. The value found is displayed in the Delay Edit Box; reopening the Settings Dialog after the measurements has been taken allows to you to view the automatically chosen delay time.

Impedance Settings

Set how Impedance is calculated and displayed. When taking impedance measurements refer either to the **Internal** impedance mode or to **QC Box Select** (the hardware setting of the QC Box determines directly the Impedance Mode, refer to 4.6).

Ohm Right Scale

If selected the impedance scale will be activated on the right of the graph. When a stereo measurement is taken, then channel B measures impedance sensing current from a dedicated QCBox ISense output.

Distortion Settings

R&B Enabled

Enables Fast-Track Rub&Buzz calculation. Available only in QC software version.

THD Enabled

Enables THD and Harmonics calculation.

% Reading

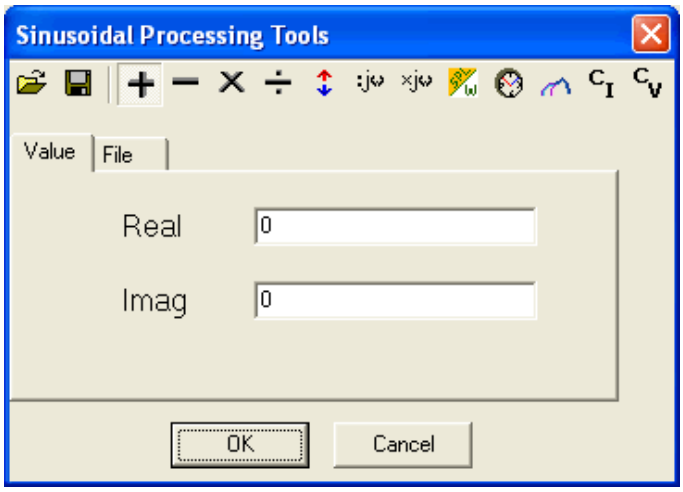
Sets distortion curves unit to % instead of dB.

Rise [dB]

Value, in dB, used by the graphical routines to raise the display of the distortion curves. Only the display of the curve is affected. The marker readings continue to display the real value which, if the rise value is different than 0, differs from the curve position.

11.2.4 SINUSOIDAL POST PROCESSING TOOLS

The POST PROCESSING Dialog gives access to very powerful tools that, once defined, can be saved, reloaded and automatically be applied to every executed measurement.



Loads a Sinusoidal process.

Saves a Sinusoidal process.

Adds a data value or compatible file to the current measurement.

Subtracts a data value or compatible file to the current measurement.

Multiplies the current measurement by a data value or compatible file.

Divides the current measurement by a data value or compatible file.

Shifts the current measurement by a dB value.

Multiplies the current measurement by complex frequency.

Divides the current measurement by complex frequency.

Uses a reference measurement file taken at speaker terminals to calculate 1m sensitivity in dBSPL/W. The reference file should have dBV Y units while the one in memory should be in dBSPL.

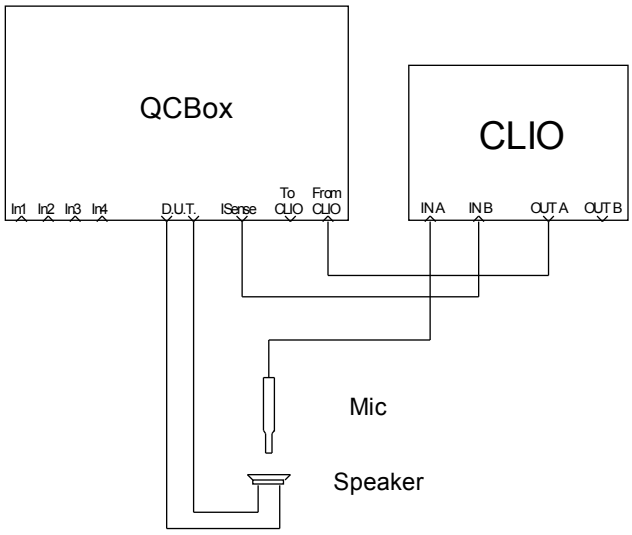
Temporally shifts the current measurement by a ms value. Affects phase response.

Merges the current measurement with the part below the selected transition frequency of a selected compatible file.

Combines the current measurement and the selected file to obtain a constant current impedance measurement. Both files should be in dBV.

Combines the current measurement and the selected file to obtain a constant voltage impedance measurement. Both files should be in dBV.

11.3 HOW TO MEASURE A SIMULTANEOUS FREQUENCY AND IMPEDANCE RESPONSE OF A LOUDSPEAKER



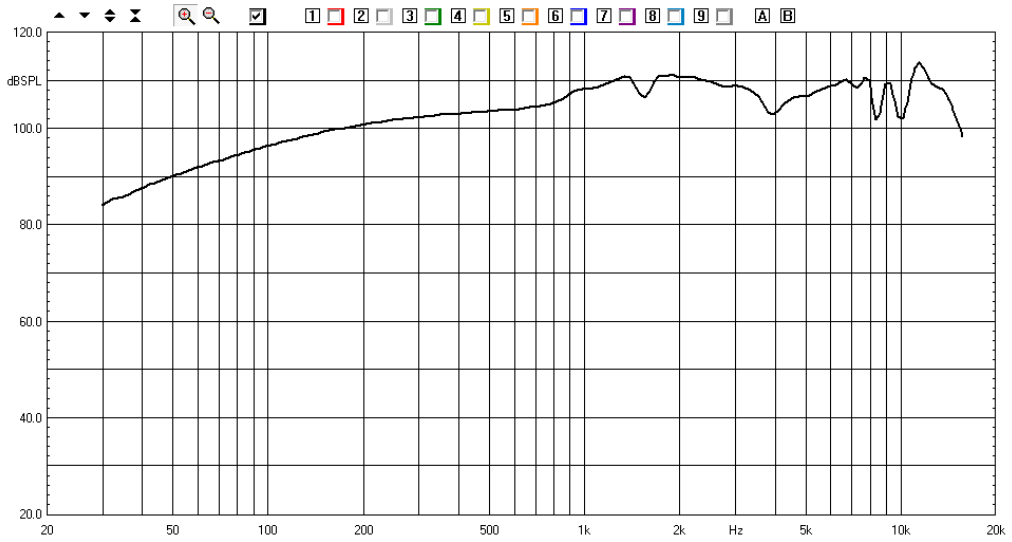
Using the ISense current sensing output of a QCBox it is possible to simultaneously measure frequency response and impedance of a loudspeaker; this tutorial will guide you through the steps nedeed while setting up this test.

11.3.1 SETTING UP THE FREQUENCY RESPONSE

Open the sinusoidal menu. Let's start with the acoustic frequency response; open the settings dialog to set up the required sweep. The main parameters affecting sweep are: frequency range, here chosen from 30Hz to 15kHz, resolution of 1/12 of octave, supposed to be fine, and speed that is set to "Normal".

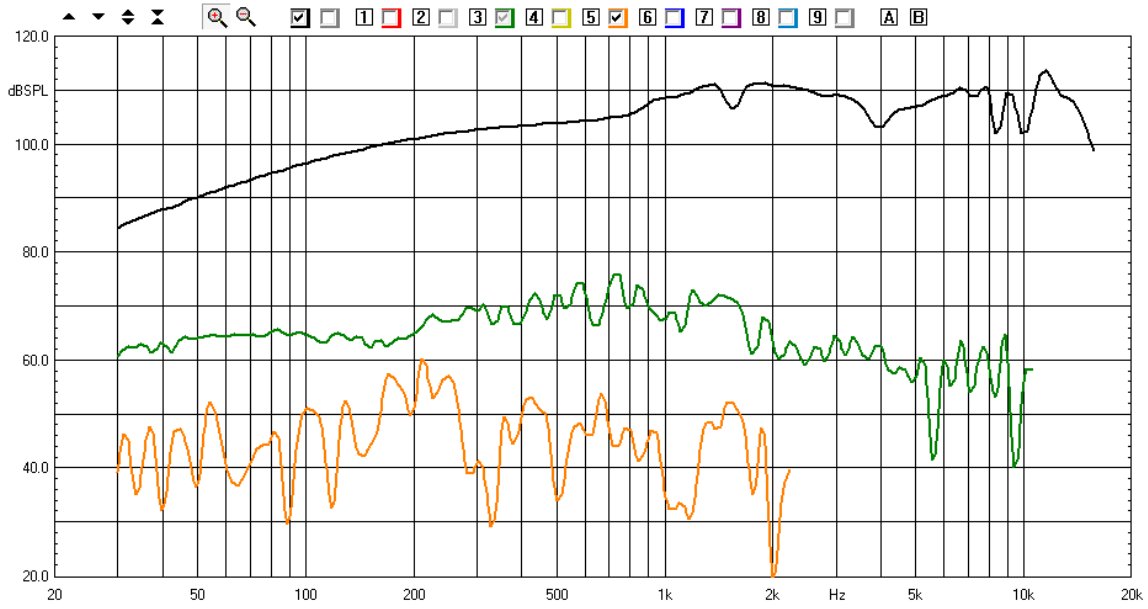
Before taking the first measurement you still need to set the proper output level (here chosen 1V at speaker terminals) as indicated by DUT specifications and accordingly set input sensitivity of CLIO input A; as the final measurement will be stereo operate separately the two input channel controls releasing the Link Input Controls button in the hardware toolbar; initial input A sensitivity is -10dBV (channel B is left to 0dBV).

Now choose CHA input channel selection and dBSPL as Y scale unit. Press go. The first measurement gives you the following result



one important parameter now clear is the sweep time that is shown in the sinusoidal menu status bar: with these settings we have 1.05 seconds sweep time. Consider it fine. Save the result to "response.sin" file.

The test should now be tuned up to take into account the acoustic environment and completed with missing settings. Open the sinusoidal settings dialog; proper delay should be set to compensate for microphone distance to loudspeaker, this may be evaluated by the two common ways CLIO gives you i.e. taking a trial sinusoidal measurement with auto delay active or taking a parallel MLS&LogChirp measurement and inspecting the impulse response; in our case we found a 0.2ms delay to be compensated, due to a quasi near field measurement with a microphone to DUT distance, in the acoustic fixture, of circa 7cm. Final settings you may require are about distortion curves; we may activate THD calculations clicking on "THD Enabled", the Rise parameter is set to 0dB as we are going to accommodate all displayed curves inside one single 100dB Y scale graph. Execute the measurement with final frequency response settings.



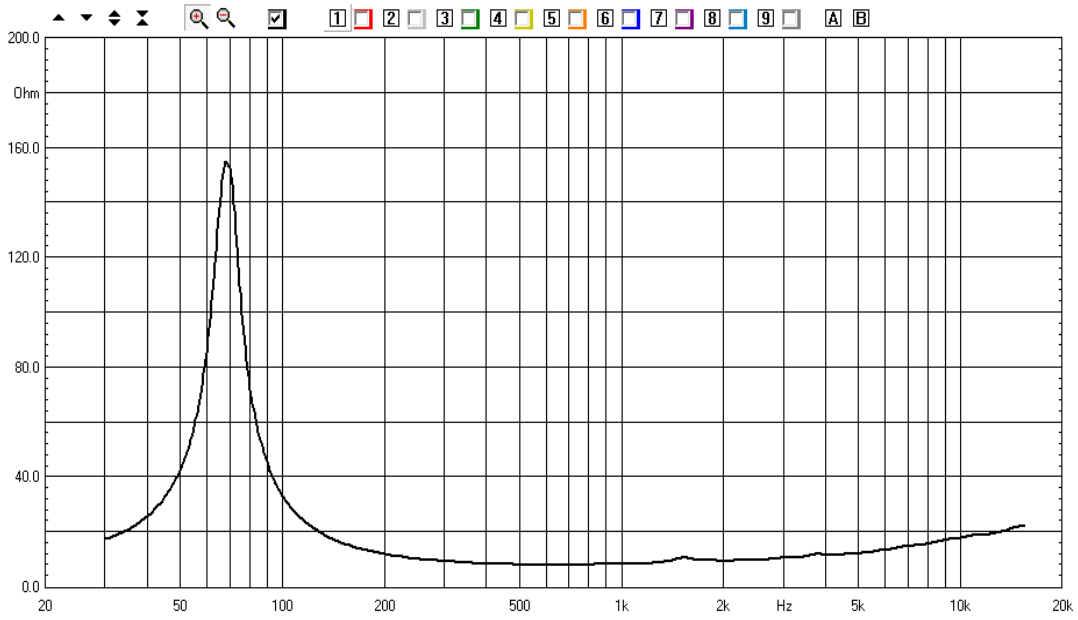
After the measurement is done we may inspect THD and 10th harmonic pressing the THD button and selecting the proper harmonic with the drop-down, in figure they are shown as overlays (green THD, orange 10th harmonic). Repeat the measurement until fully confident with the results obtained, eventually refine the settings as needed.

11.3.2 SETTING UP THE IMPEDANCE RESPONSE

We put now our attention to the impedance response of our loudspeaker.

We choose CHB with the input channel selector and Ohm as Y Scale unit; inside the sinusoidal settings dialog leave all previous settings unchanged as they will accompany us to the final reference measurement; only change the impedance settings to "QCBox Select" to reflect QCBox operation.

As the output level has already been set for the acoustic test we only have to deal with input sensitivity for channel B; a settings of -30dBV or -40dBV is usually correct for ISense impedance tests. The measurement looks as follow.



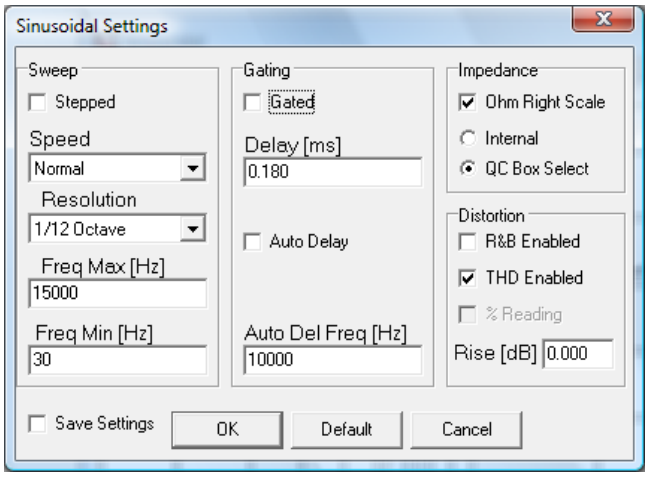
Save the result to "impedance.sin" file.

11.3.3 INTEGRATING THE TWO-CHANNELS MEASUREMENT

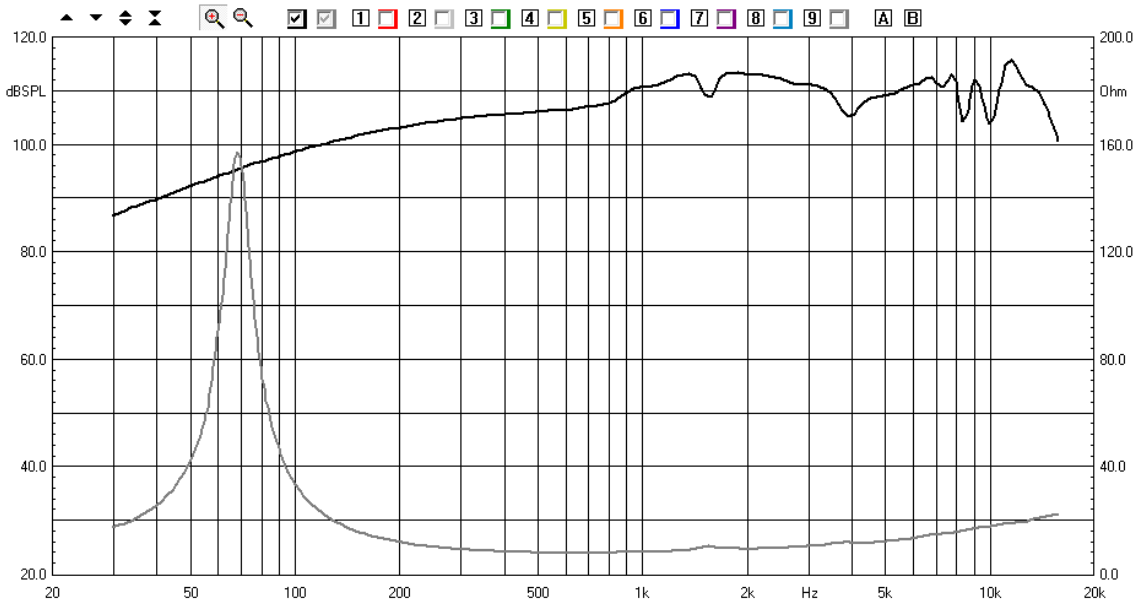
Starting from the actual situation, i.e. having just measured impedance relying on settings that accumulated from the previous frequency response measurement, we are now ready to integrate all of our work to realize a single stereo sinusoidal measurement.

Have the impedance measurement loaded in memory; select CHA&B with the input selector, change the Y Scale unit to dBSPL; CLIO is now ready to take a two channels measurement with main unit set to dBSPL; as the measured unit for channel B needs to be Ohm we must open the sinusoidal settings dialog and select "Ohm Right Scale": **in this way channel B will measure impedance using the right scale to identify it**.

The final sinusoidal settings are:



Press Go; the graph obtained has frequency response measured from channel A and refers to left scale while impedance response comes from channel B referring to right scale. Note that the two curves displayed are measured and controlled by dedicated checkboxes, no overlays are active.



The measuremenet can be saved as “response_impedance.sin”. To properly set scales it is useful to directly input values at their extremes; refer to 6.2 and 6.4 for details about this.

Read carefully 6.2.1 about the stereo measurement display features.

11.4 A BRIEF DESCRIPTION ON SETTINGS EFFECTS

11.4.1 STEPPED VS. NOT STEPPED

Although measuring speed increases, use of a "not stepped" sweep can adversely affect measuring results in several circumstances. As an example that should make this clear, let's see what happens while measuring the impedance of a woofer in Internal or Constant Current Mode. Please refer to Measuring Impedance for more information on this topic. In both conditions the loudspeaker is driven from a high impedance source and its damping is only mechanical. Fig.11.1 shows a 6" woofer, driven by a 100Ohm output impedance generator delivering a sinusoidal burst 200ms long, at its resonance frequency. When the excitation stops, the device continues to move, and therefore produce back electromotive force (EMF) voltage for more than 50 ms. Something very similar happens at start-up. In this situation, if CLIO is set in "not stepped" mode, it will acquire this EMF together with the actual results.

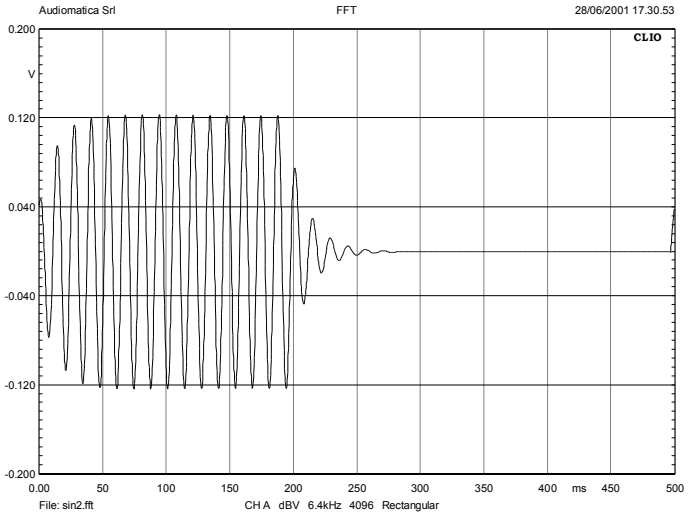


Figure 11.1

Fig.11.2 shows the same driver impedance taken both in stepped (black) and "not stepped" mode (red). The "not stepped" curve is simply wrong. Conclusion: use always stepped mode or the highest frequency resolution if the behavior of the device to be measured is unknown.

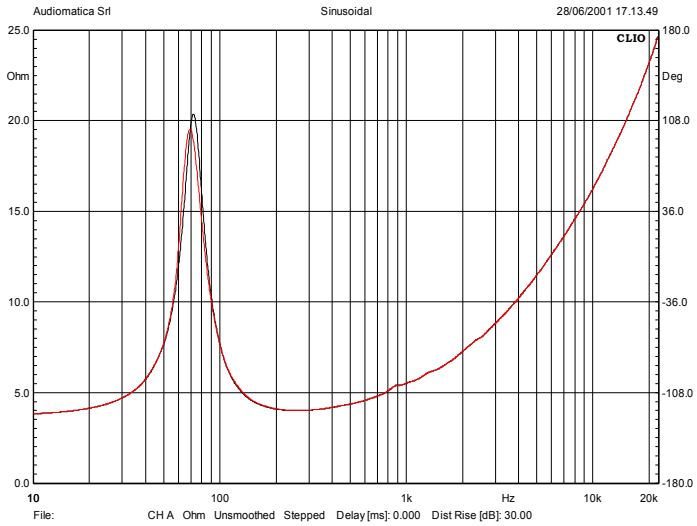


Figure 11.2

11.4.2 FREQUENCY RESOLUTION

Here the lower the resolution the faster the measuring time. Impedance measurements are again a powerful way to explore problems. Fig.11.3 shows two impedance measurements taken from the same 16" woofer with 1/24 octave resolution (red) and 1/6 octave resolution (black). Deriving T/S Parameters from the black curve would lead to serious errors. This is an extreme case, a huge woofer with high Qms. Different curve shapes can sometimes be accurately quantified even with 1/3 octave resolution.

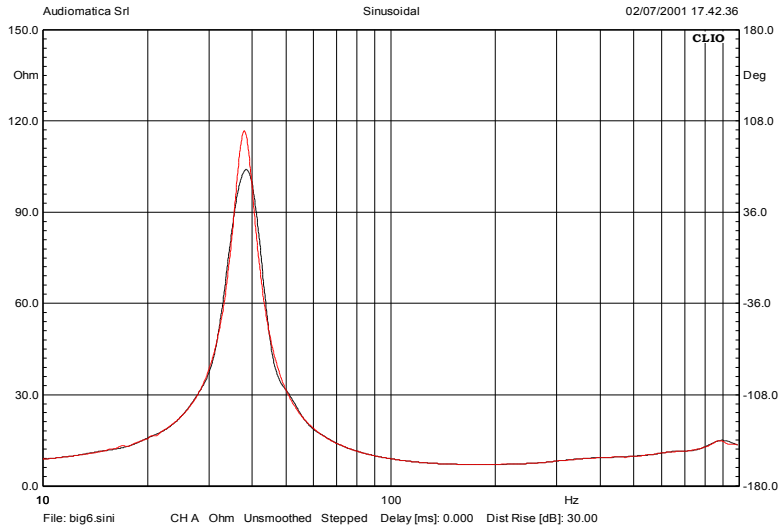


Figure 11.3

11.4.3 GATING

Enabling Gating allows quasi anechoic Frequency Response to be carried out in normal environments, with obvious and less obvious limitations. Regarding the geometrical environment required, Sinusoidal analysis does not differ from what has been said about MLS. Nevertheless the latter gives a much more intuitive approach. It is strongly suggested that you become very familiar with quasi anechoic measurements using MLS before dealing with Gating. What follows is a brief description of the parameters involved. We are going to use some figures to help our understanding. Fig.11.4, repeated from the MLS chapter, shows a typical setup, while Fig.11.5 shows what really happens, using as stimuli a 20ms 1kHz Sinusoidal Burst.

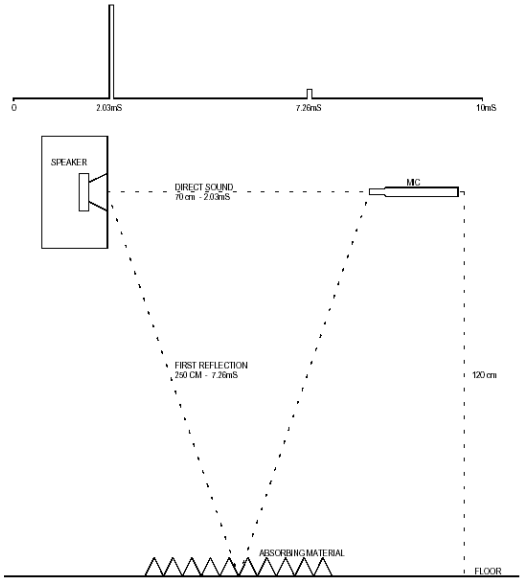


Figure 11.4

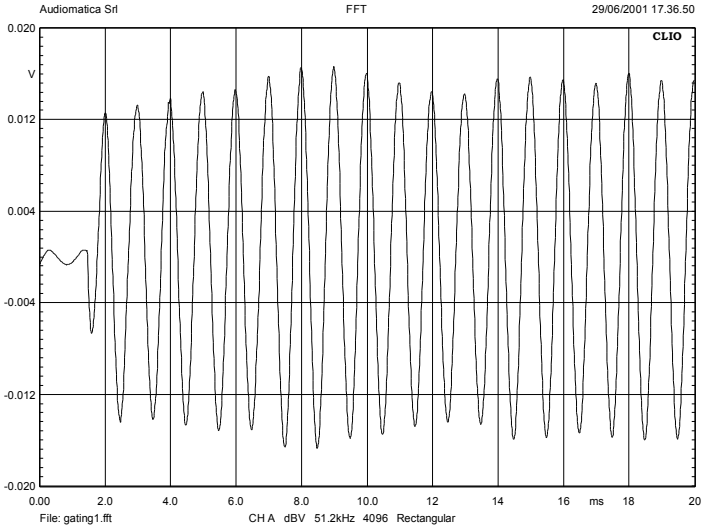


Figure 11.5

The time of the sound propagation delay is clearly visible at the beginning, roughly 1.3ms. This is the value users should input in the Delay Edit Box. Thereafter is the short time that the device takes to start. This is usually short enough to not affect amplitude evaluation but adversely affects distortion measurements. At 7 and 8 ms, where there are two consequent sinusoid positive peaks, an increase in level is clearly visible. This is the effect of the first reflection (floor); as it affects amplitude

evaluation it should therefore have been stopped before it is processed. The time the system evaluates the signal is usually defined as Meter On time. This is automatically set by CLIO around the value of 6 ms, as long as the Frequency involved is high enough to allow this. Fig.11.6 is a plot of the Meter On Time Vs Frequency CLIO uses.

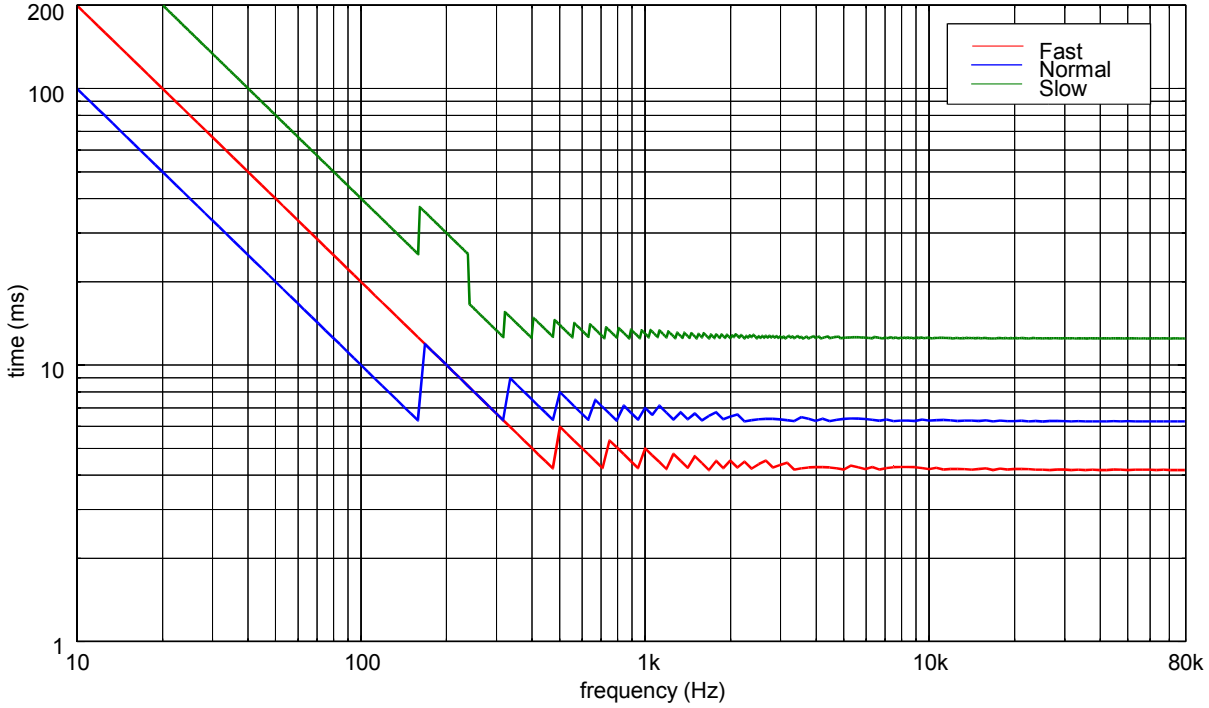


Figure 11.6

Users should use this graphic to determine the lowest Frequency that has been measured in anechoic state. Using a fixed predefined Microphone and Loudspeaker location makes all these parameters easier to define in routine measurement processes. But for new situations it's very advisable to run an MLS and have a look at the impulse response obtained. Fig.11.7 shows the labels of the corresponding data to be input as gating parameters.

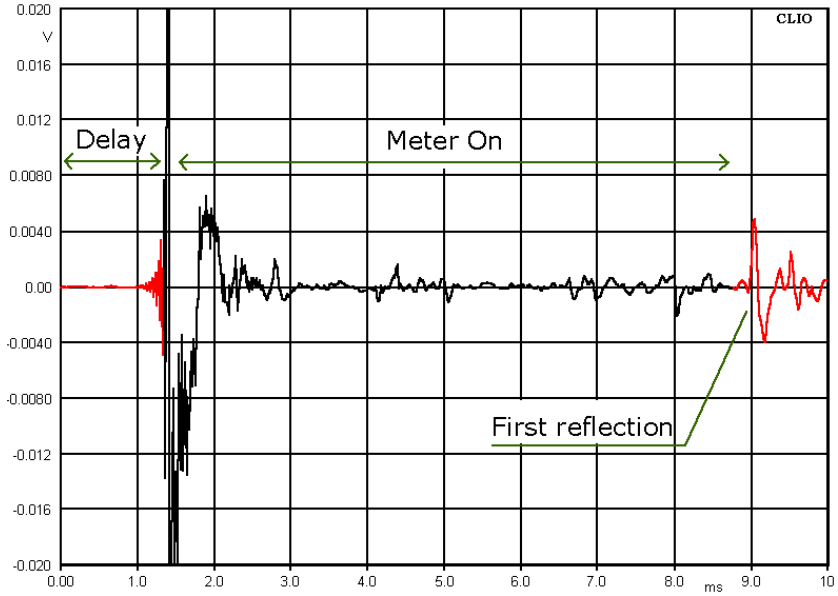


Figure 11.7

11.5 DISTORTION AND SETTINGS

Sinusoidal stimuli allow CLIO to evaluated distortion in its single harmonic form. If not Set in Impedance Mode, CLIO always evaluates harmonics from second to tenth plus THD and allows the display of each one separately via the selection drop-down. While it is simple to obtain meaningful distortion figures of electrical devices, measuring Loudspeaker distortion in normal environments (without anechoic chamber) is not easy. We will only give some advice here, relying on examples, as the topic is far beyond the scope of this User Manual. To do this we will use CLIO's FFT Menu in quite an advanced way. Distortion evaluation is adversely affected by several parameters, two of which are the most important.

Noise

50 dBSPL of ambience noise, a common figure, usually does not affect Amplitude evaluation which is usually carried out at an average level of 90dBSPL. This is particularly true using CLIO Sinusoidal Analysis capability which, by means of DSP filtering, allows exceptional S/N Ratio. Unfortunately evaluating 1% distortion means looking for signals that are 40dB lower than the 90dBSPL mentioned above, in the same order of magnitude as environment noise.

Gating Effects

Device settling time, non perfect delay removal and reflections arriving within the sampling time (Meter On) seriously affect distortion measurements, creating artifacts.

It is advisable to perform a noise evaluation using FFT Analysis. Fig.11.8 shows two curves, the first (red) is obtained with the Max Hold function, the second (black) with Min Hold.

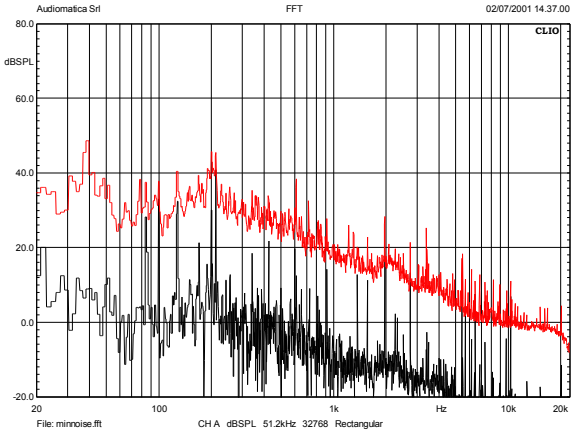


Figure 11.8

The first, a sort of worst case, should be taken as the reference lower distortion floor once raised 10dB at least. The second is useful to identify pure tones in the noise spectrum; these are likely to produce both distortion increase as well as cancellation artifacts. In our case these tones are caused by more than 20 different computer fans, spread everywhere in the room. Supposing we carry a distortion analysis at an average 90dBSPL, residues below 1% are difficult to evaluate up to 600Hz, things are much better at higher frequencies. Remember that the frequency axis should be referred to the harmonic we are looking for, not to the fundamental. The obvious solution to overcome noise is to increase the level. One way is to put more voltage at the Loudspeaker terminals; unfortunately this increases distortion by itself even if it provided important information regarding the device. The second

way is to narrow the Microphone to Loudspeaker distance. The next figures, dealing with Gating Effects, refer to a Microphone at 11.5cm (4.5") in front to a good quality tweeter. FFT size is set to 512 points, the equivalent of about 10ms Meter On at 48000Hz sampling rate. Fig.11.9 shows the effects of a wrong delay in capturing a 2kHz 10ms tone burst. All harmonics are buried below the effects of this wrong setting.

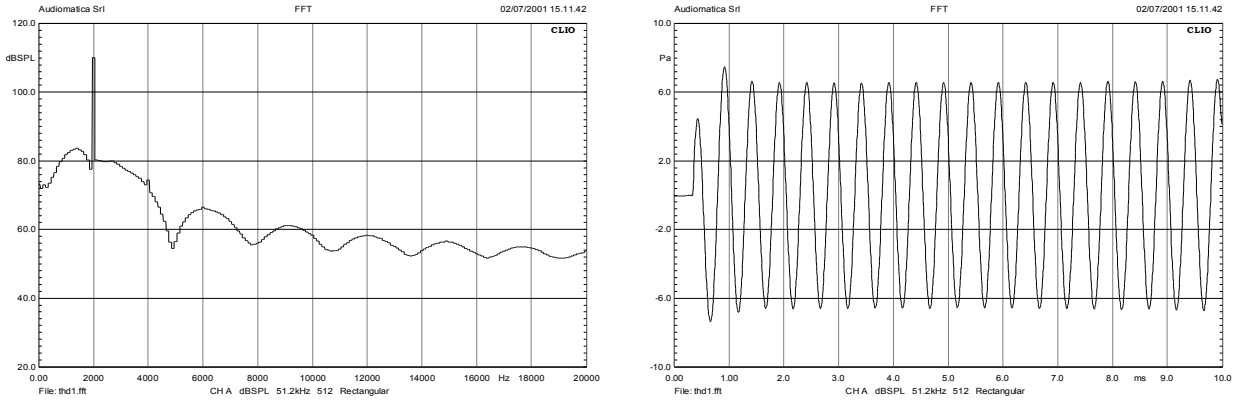


Figure 11.9

Fig.11.10 shows the effects of the device settling time as the delay is now correctly set to 0.35ms. 40dB down the harmonics (1% distortion) should be visible now. As the tweeter performs better than this what we see is the second harmonic canceling the broad spectrum caused by the device settling time.

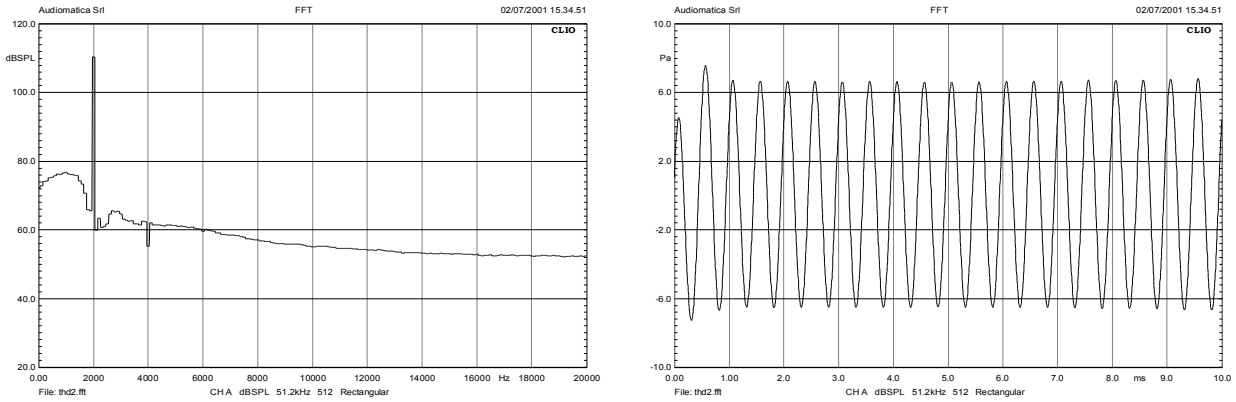


Figure 11.10

Fig.11.11 shows the spectrum when the delay has been set to 1.5ms. The third harmonic, 64dB (0.06%) below the fundamental is clearly visible.

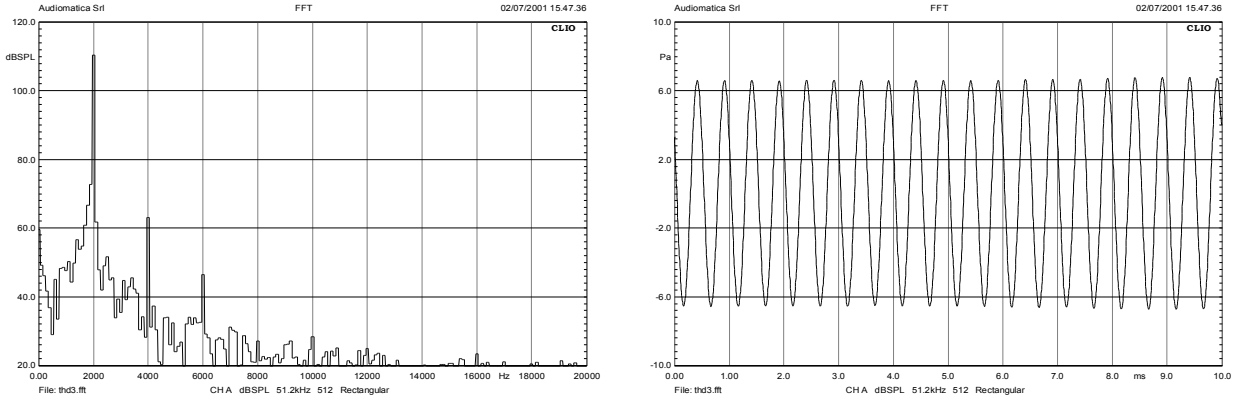


Figure 11.11

Finally Fig.11.12 shows the distortion analysis carried out with the same microphone distance as in the past examples and the gating delay set to 1.5ms with the auto delay option disabled. Fundamental is red, second harmonic (+30dB) blue and third harmonic (+30dB) green.

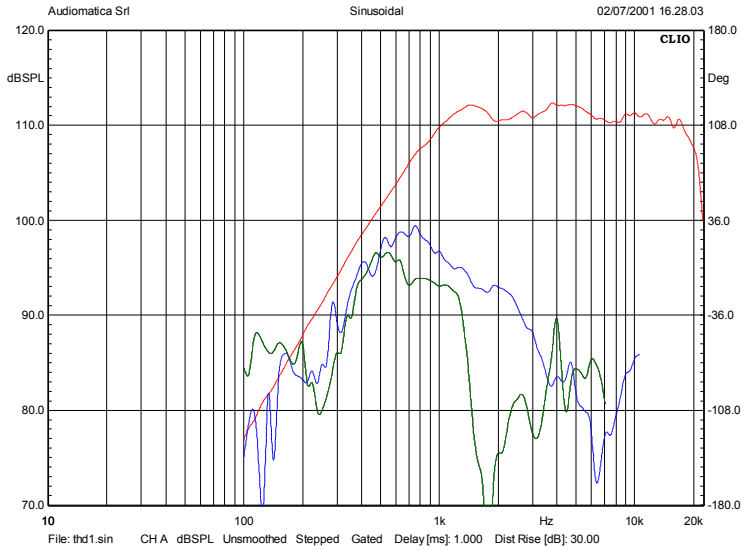


Figure 11.12

12 WATERFALL , DIRECTIVITY & 3D

12.1 INTRODUCTION

The Waterfall, Directivity & 3D post processing routines **(3D post-processing is available only with QC version software)** give CLIO the possibility of making 3-D or Color plots by adding a third dimension (time or degrees) to classical amplitude-frequency graphs and to visualize and export 3D polar response balloons.

Waterfalls are used to characterize the anechoic sound decay of a loudspeaker or the sound decay in a room.

The Waterfall post processing permits the following 3-D or Color types of analysis:

- **Cumulative spectral decay (CSD)**
- **Energy Time Frequency (ETF)**

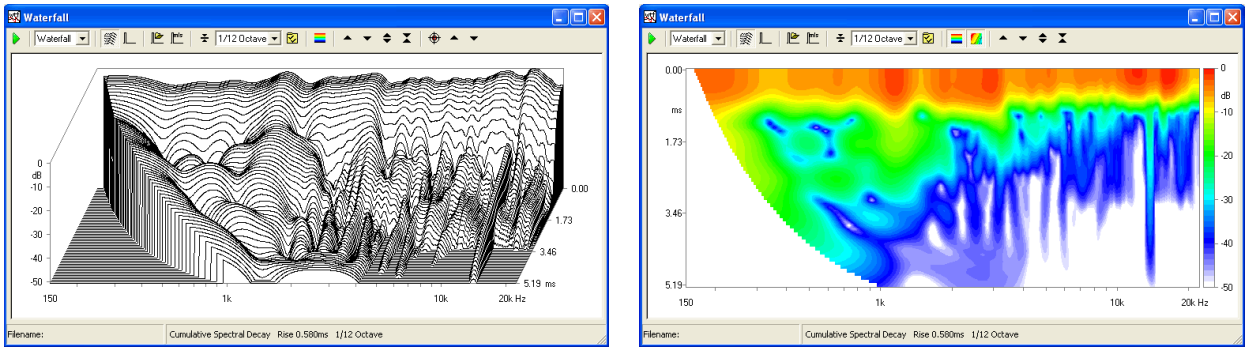


Figure 12.1

Directivity analysis characterizes the radiation of a loudspeaker versus vertical or horizontal angle.

The Directivity post processing permits the following analysis:

- **3-D directivity (waterfall like)**
- **Color map directivity**
- **Classical polar plots**

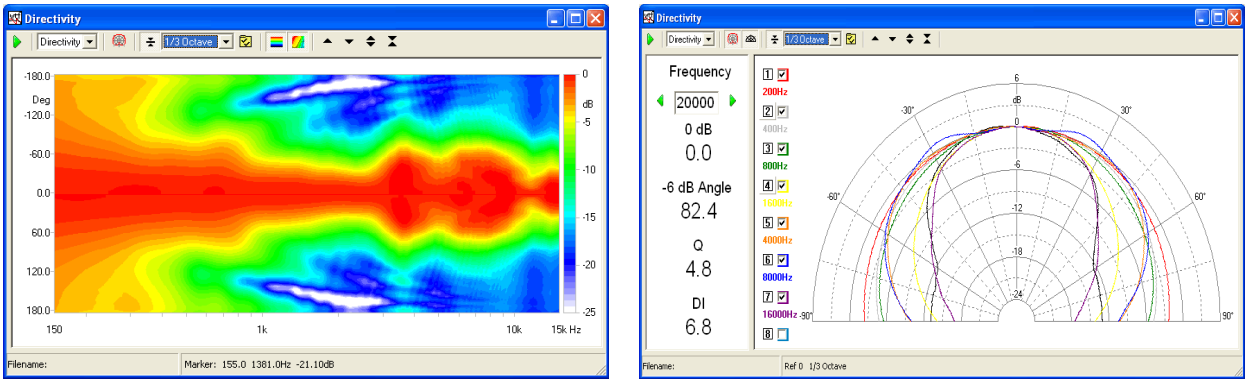
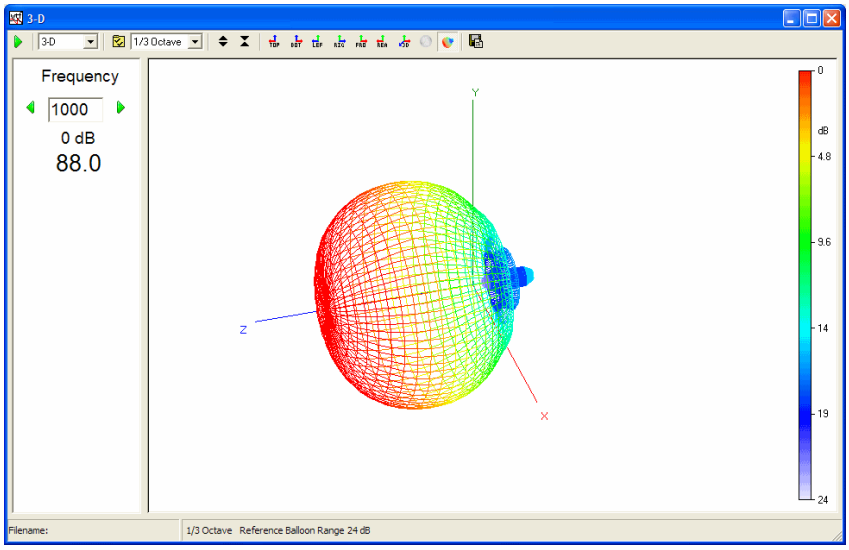


Figure 12.2

The 3D post processing permits the following **(QC version software only)**:

- **3-D balloon at standard 1/3rd octave frequencies**
- **Ballon Export to EASE or CLF**



12.2 WATERFALL, DIRECTIVITY & 3D CONTROL PANEL

Fig 12.1 and 12.2 show the Waterfall, Directivity & 3D control panel in many of its possible configurations; as you may imagine the post processing capabilities of this menu are very powerful.

It is important to understand which is **the source of data** for the waterfall and directivity analysis.

Waterfall
A waterfall analysis is a post process applied to a measured **impulse response**. Please refer to chapter 10 (MLS&LogChirp) to have details on how to measure an impulse response.

Directivity
A directivity analysis is a post process applied to a **set of measured frequency responses**. Please refer to chapters 9,10 and 11(FFT, MLS&LogChirp and Sinusoidal) to have details on how to measure a frequency response.

3D
A 3D analysis is a post process applied to a **set of measured MLS&LogChirp frequency responses**. Please refer to chapter 10 (MLS&LogChirp) to have details on how to measure a frequency response.

12.2.1 COMMON TOOLBAR BUTTONS AND DROP DOWN LISTS

Starts a waterfall, directivity or 3D calculation.

Analysis drop down
Selects Waterfall, Directivity or 3D analysis.

Smoothing drop down
Selects the smoothing factor in fractions of octave.

Enters the settings dialog. See 12.3.1 and 12.5.1.

12.3 WATERFALL SPECIFIC CONTROLS

If pressed the waterfall spectra will be referenced to the rearmost one; the directivity spectra will be referenced to the one identified by the Z-Ref value (see 12.5.1)

Displays a color map instead of 3D plot.

Interpolates colors in order to obtain smooth level contours.

Moves the plot up.

Moves the plot down.

Expands the plot changing its Y scale. The Y range is reduced.

Compresses the plot changing its Y scale. The Y range is increased.

When pressed the waterfall plot is displayed.

When pressed the impulse response loaded in memory is displayed.

Loads an impulse response from disk (from MLS binary files).

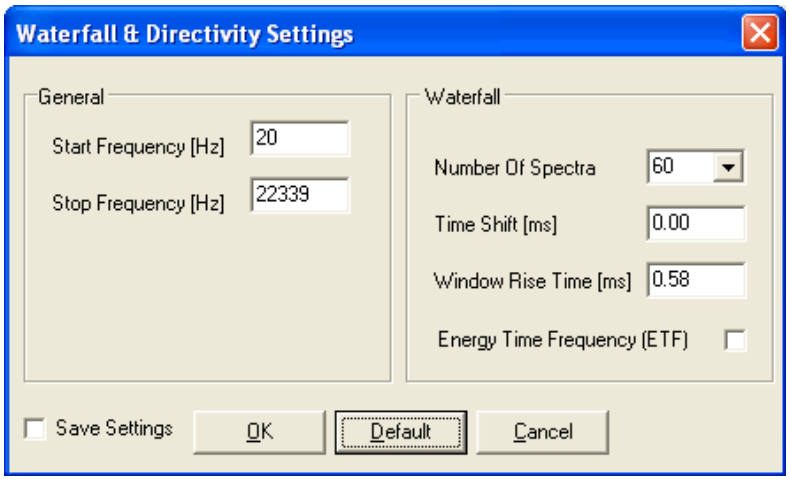
Takes current MLS impulse response for waterfall calculation.

Enters the Marker mode (see Fig. 12.5). In this operating mode it is possible to inspect the single data points of each waterfall slice by clicking and dragging the mouse.

When in marker mode, moves the selected waterfall slice backwards. The same action is obtained with the **Up-Arrow** keyboard shortcut.

When in marker mode, moves the selected waterfall slice frontwards. The same action is obtained with the **Down-Arrow** keyboard shortcut.

12.3.1 WATERFALL SETTINGS



Start Frequency

Selects the start frequency for the analysis.

Stop Frequency
Selects the stop frequency for the analysis.

Number of Spectra
Selects the number of data slices to display.

Time Shift (ms)
Selects the time between two consecutive spectra.

Window Rise Time (ms)
Selects the rise time of the data selecting window. Valid only for CSD.

Energy Time Frequency (ETF)
Selects ETF mode waterfalls.

12.3.2 WATERFALL OPERATION

As already stated, the data source for a CSD or ETF waterfall is a measured impulse response

Once you have loaded an impulse response inside the Waterfall control panel you may easily inspect it, in the same way you also do with the MLS Impulse control panel (see chapter 10). Of great importance is to select the start time and stop time of the analysis: **start time**, the Start Window value selected in the impulse response represents time zero for the waterfall; **stop time**, the Stop Window value selected in the impulse response represents the last processed CSD slice unless a different Time Shift has been selected.

CSD (**Cumulative Spectral Decay**) is intended primarily for anechoic loudspeaker evaluation; in this case only the data between the start and stop time is analyzed; each successive slice considers time data from its relative start time (the rearmost, at time zero, has start time equal to the start window of MLS) to the fixed stop time, the data being windowed by a particular time window with a smoothed rising edge (see literature for a discussion about this). Normal values for the Window Rise Time lie within 0.1 and 0.6ms. In CSD mode, should the Time Shift value be left at zero, the routine will automatically calculate it, spacing the selected Number of Spectra in the interval defined by start and stop times; if Time Shift is forced by the user be sure to set it small enough to permit the last spectra to be calculated; if the fixed stop time is passed, then the calculation defaults as in case of zero Time Shift.

When representing a CSD the program automatically hides the low frequency part of the spectra that has become unreliable due to the time-frequency uncertainty principle.

ETF (**Energy Time Frequency**) is intended for room acoustic evaluation; in this case all MLS data starting from the start time are computed; then, successive slices are calculated moving their initial point of the Time Shift value (see 12.3.1 the Settings dialog).

12.4 MAKING A CUMULATIVE SPECTRAL DECAY

A cumulative spectral decay starts loading an impulse response from disk. Suppose we have taken an anechoic response of a medium sized two ways loudspeaker; the impulse response is shown in Fig.12.3.

Let's first select a reflection free part of it. By selecting the start and stop window points we obtain the first two information parameters required for the waterfall facility: zero time will be referenced to the start of the start window, while the Z axis will provide the measurement range between the stop and start window points (unless a Time Shift is chosen).

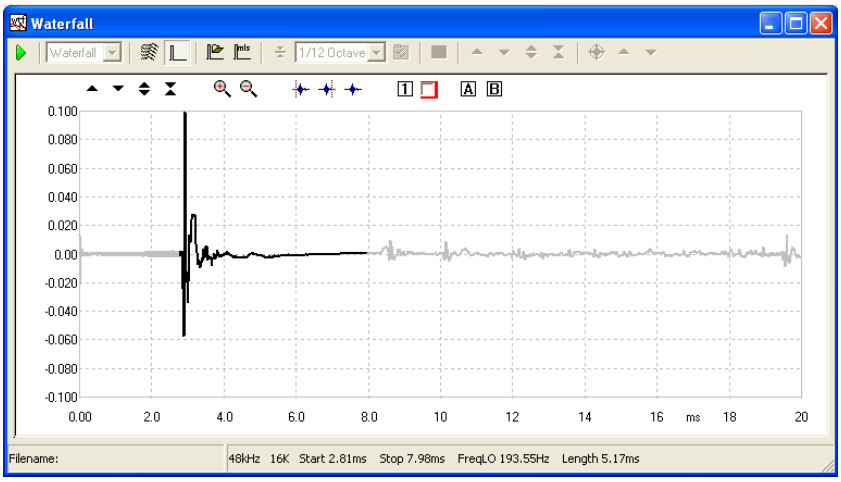


Figure 12.3

Going inside the Waterfall Settings dialog we decide to view our measurement between 150 and 20000Hz, then apply 1/12 octave smoothing. We are now ready for a waterfall!

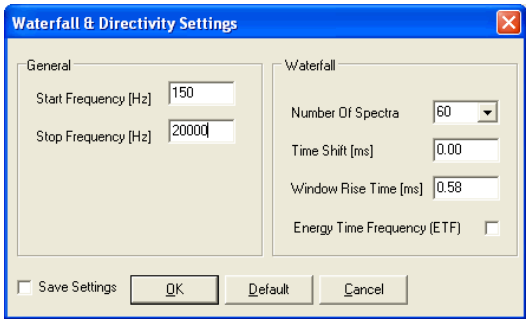


Figure 12.4

The Go button is enabled. Press it, you should obtain a waterfall like the one in the left part Fig. 12.5. Press now the Color Scale button followed by the Interpolate Colors buttons, now you should have the color map shown in the right part of Fig. 12.5. The two representations are not exclusive, they mutually complement each other; you will gain experience understanding all the subtle details of a waterfall processing and how they are represented either in the 3D or in the color map. For example the color map represents better the frequency of decaying modes as they result as straight color patterns parallel to the time axis; the 3D waterfall is more familiar when you look at zero time frequency plot and try to visualize how it modifies during decay.

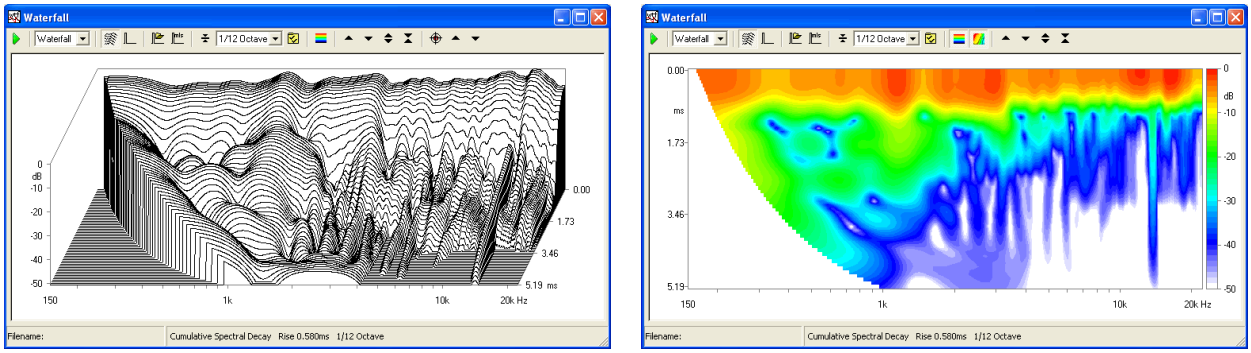


Figure 12.5

One powerful way to inspect a waterfall is to enable its marker. Press the button. The display should change as in Fig.12.6. It is very easy to locate frequency zones where the decay 'suffers', like the peak around 2200Hz. After placing the cursor on it, it is possible to quickly move back and forth the calculated slices by means of the up and down keyboard arrows.

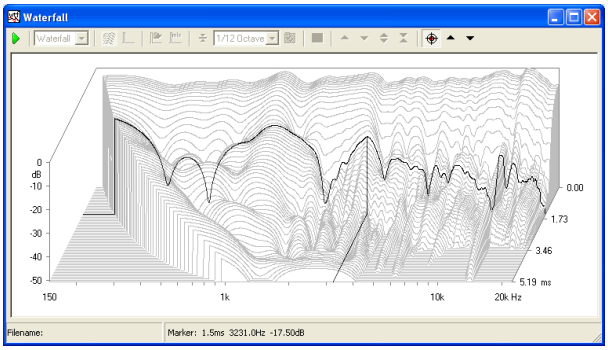


Figure 12.6

Let's now change the CSD aspect. Go to the waterfall settings dialog and input 0.1ms Time Shift. After recalculating it you obtain the plot in Fig.12.7; as you now notice the slices are closer in time and decaying modes are more evident. Since the time span chosen for this waterfall was about 6.1ms the maximum allowed Time Shift you could input was around 0.2ms (considering 30 spectra as in this case).

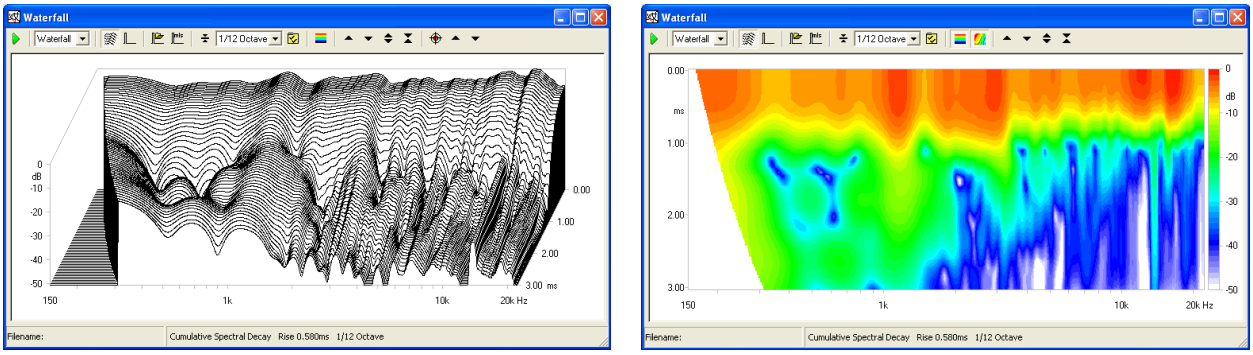


Figure 12.7

Let's now change the CSD aspect again. Go to the waterfall settings dialog and check the Reference box. After recalculating, you obtain the plot in Fig.12.8

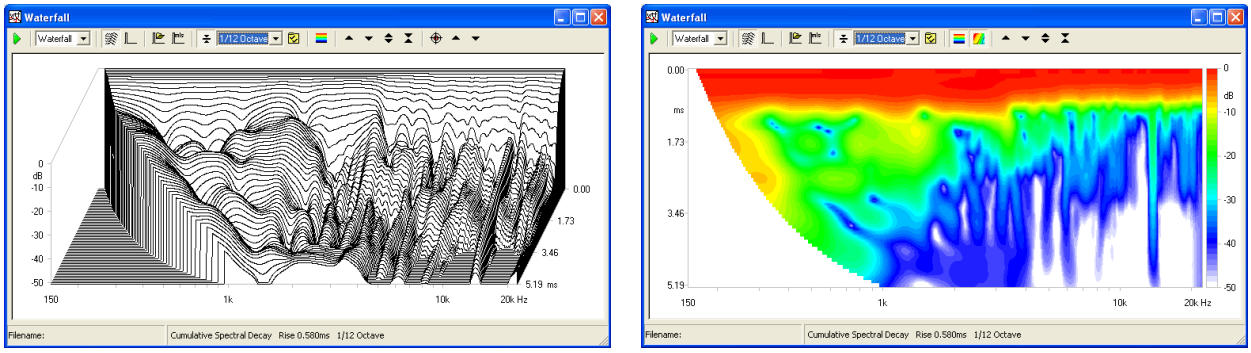


Figure 12.8

As you can now see the slices are referenced to the first one (the rearmost); thus allowing decays of different frequency regions to be compared more easily.

Now change the Windows Rise Time from the default 0.58ms to 0.1ms and recalculate the CSD. The result is given in figure 12.9.

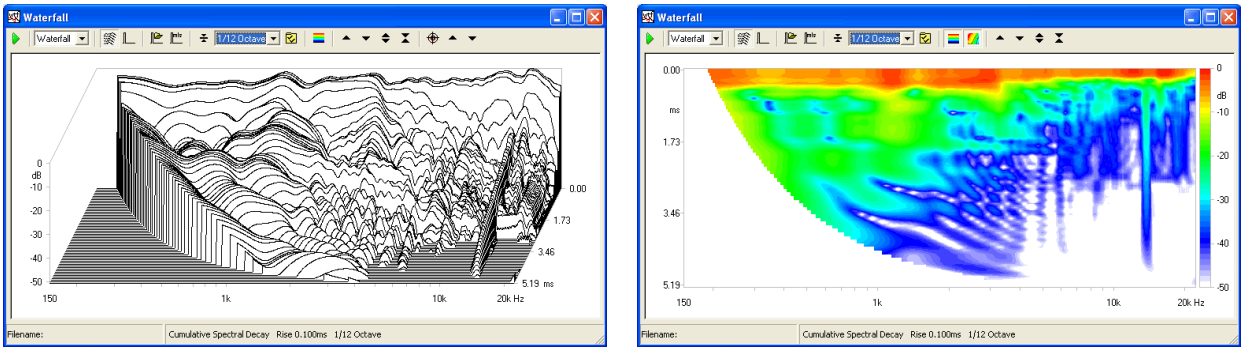


Figure 12.9

12.5 DIRECTIVITY SPECIFIC CONTROLS

If pressed the directivity spectra will be referenced to the one identified by the Z-Ref value (see 12.5.1)

Displays a color map instead of 3D plot.

Interpolates colors in order to obtain smooth level contours.

Moves the plot up.

Moves the plot down.

Expands the plot changing its Y scale. The Y range is reduced.

Compresses the plot changing its Y scale. The Y range is increased.

Enters the polar pattern mode.

Mirror data

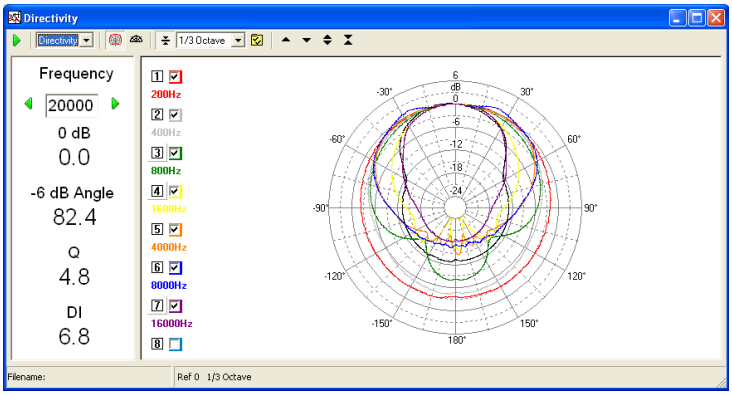


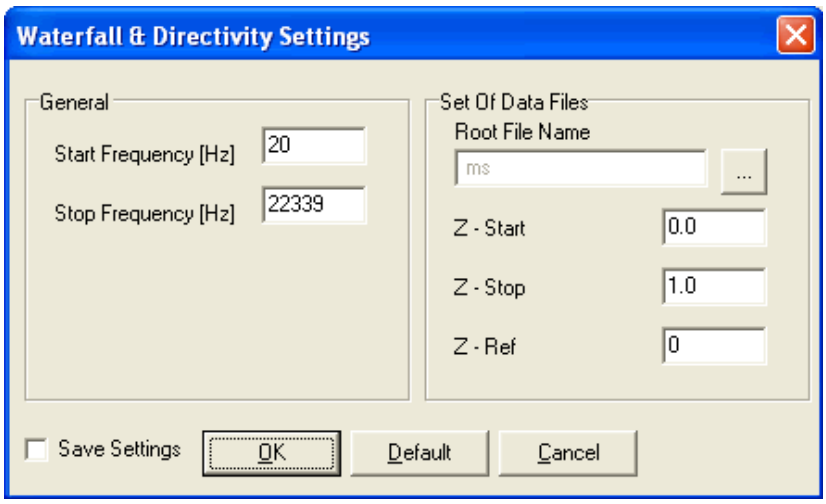
Figure 12.10

When in polar pattern mode moves analysis frequency up 1/3 of octave.

When in polar pattern mode moves analysis frequency down 1/3 of octave.

Selects an half space polar pattern. Refer to the right Fig.12.2.

12.5.1 DIRECTIVITY SETTINGS AND OPERATION



Start Frequency
Selects the start frequency for the analysis.

Stop Frequency
Selects the stop frequency for the analysis.

Root File Name and browse button
The name of one file within the set to be displayed. By pressing the associated button it is possible to browse the disk and choose the file.

Z-Start
Value associated to the first (rearmost) file.

Z-Stop
Value associated to the last (foremost) file.

Z-Ref
Value associated to the file to be taken as reference.

The radiation characteristics of a loudspeaker or driver versus frequency and angle rely on a huge amount of data i.e. a set of frequency responses (taken at different angles on the vertical or horizontal planes) saved to disk; for example it is common to work with sets of 72 files representing the frequency response taken at 5 degrees angles to represent a complete rotation in a plane.

With the directivity analysis you get a powerful way for synthesizing a large number of measurements in a single color map or 3-D graph. This control panel allows the representation of the classical **polar response** of a loudspeaker as in Fig. 12.10.

In order to identify the set of files it is important that all of their names follow a particular syntax, that gives certain information to the processing routines.

The syntax follows: **<NAME><UNITS><VALUE/100>.MLS.**

NAME is a common file name, UNITS are the common measurement units (to be displayed in the graph as Z axis label) and VALUE is a unique value identifying the single file; these quantities needs to be separated by **spaces**, it is possible to give negative numbers to VALUE. For example '**mydriver deg -250.mls**' is a valid file name: as the name tells it is a measurement named **mydriver** with units **deg** taken at **-2.5** (250 divided by 100) units value. If the units are not specified within the name then the processing defaults to **deg**.

The auto-saving and naming capabilities of CLIO render the job of measuring and creating a complete directivity data set an easy and automatic task (see later 12.6 for an example).

It is possible to identify one file within the set as the reference with the Z-ref value; when in reference mode all calculations will then be done referenced to it.

12.6 MEASURING LOUDSPEAKER SINGLE POLAR DATA (1D MODE)

We will use a PC controlled turntable under CLIO's control, and the automation possible within the MLS control panel using the Autosave and naming rules.

Now suppose we want to measure and give a graphical representation of the polar response of the same two ways loudspeaker analyzed in 12.4.

We need to measure its anechoic frequency response, at various angles and save the files following the rules given in 12.5.1.

12.6.1 PREPARING AUTOSAVE AND THE MLS CONTROL PANEL

For this test we would like to measure the speaker from -180° to +180° (in intervals of 5°) relative to its front baffle.

The MLS measurement should be set in a particular way in order to automatically acquire the responses at various horizontal angles. **To do this we will use the autosave function, the loop mode and the link to the turntable control.**

Let's start with setting the autosave function (see also 5.3). Pressing **Alt-F2** we recall the Autosave Setting dialog (fig.12.11); here we input the desired file name ('RogersHalfChirp'), start (-180) increment (5) and total (73) values.

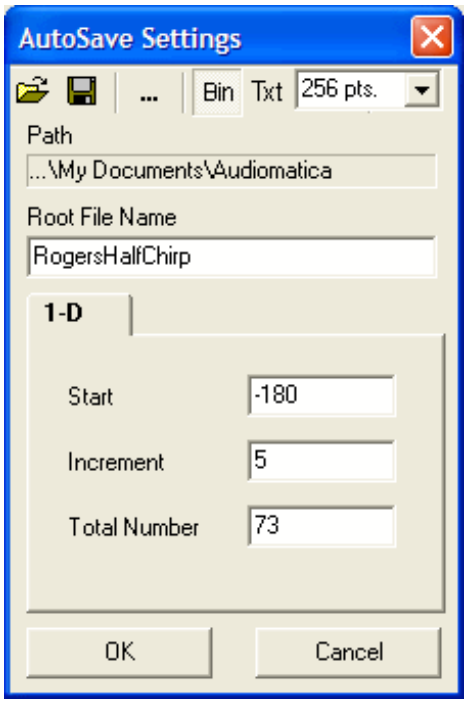


Figure 12.11

The MLS menu is now ready to start taking measurements; we only need to properly set the turntable and its control.

12.6.2 PREPARING THE TURNTABLE

Open the Turntables Control dialog and select the connected turntable (See 4.7).

Outline ET/ST (TTL Pulse)

We assume that the Outline ET/ST turntable is properly connected to your PC (refer to 4.7.1 for details). To prepare for this measurement session you need to:

1) Manually set the front selector labeled 'Deg Step' to 5°.

2) Rotate the turntable counterclockwise until you reach the desired start position: as we want to start from -180° position it at 180°.

3) Recall the Turntable Settings dialog (Fig.12.12), set Resolution at 5°, Speed at 0.75 RPM

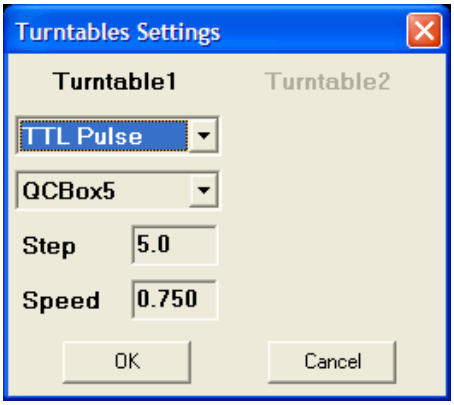


Figure 12.12

In order to **link the TTL pulse generation to measurements** leave the Turntables Control dialog open during the measurements.

Outline ET250-3D or LinearX LT360

We assume the turntable is properly connected to your PC (refer to 4.7.1 for details). To prepare this measurement session you need to:

1) Set the turntable step to 5° using the turntable specific supplied software or turntable panel hardware controls

2) Rotate the turntable counterclockwise until you reach the desired start position: as we want to start from -180° position it at 180°.

In order to **link the turntables to measurements** leave the Turntables Control dialog open and the turntable linked during the measurements.

12.6.3 TAKING THE MEASUREMENTS

You are now ready to begin the measuring session. We suggest you to take an initial measurement (with the speaker in place over the turntable) to verify all the parameters, especially viewing the acquired impulse response and setting the start and stop values of the measurement window. These values will be applied to all the measurements taken; consider, in this respect, the problem of the trajectory of the acoustic center of the speaker during the rotation.

The last thing to do is to activate Autosave and Loop; to do this we press the corresponding toolbar buttons (Fig.12.13).

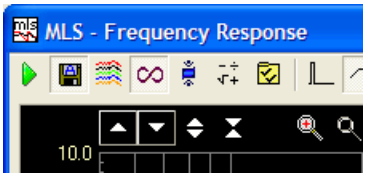


Figure 12.13

Press Go. After each MLS measurement is taken you will see the turntable rotating and CLIO waiting for a sufficient period of time to allow the turntable to stabilize before automatically taking the next measurement. Should this time be insufficient you have to reset the turntable speed value accordingly. The autosave function will refresh the file name after each measurement (Fig.12.14).

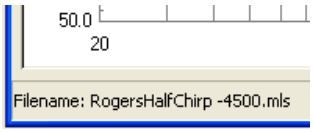


Figure 12.14

After the 73 measurements are taken the session should end while the autosave and loop buttons reset.

12.7 REPRESENTING SINGLE POLAR DATA

To represent the measured data we need to select the Directivity mode in the Waterfall, Directivity & 3D control panel. Then enter the Directivity Settings dialog and press the browse button. Entering our data directory we find the situation in Fig.12.15:

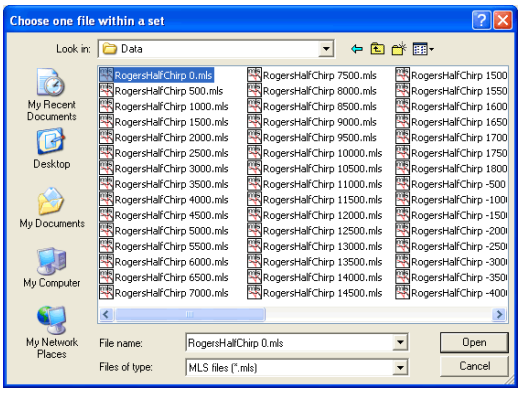


Figure 12.15

The set of files is composed by 73 files; it is sufficient to choose one of them. It is now important to identify the initial and last files for display, this is done with the Z-start and Z-stop values to be input as in Fig.12.16. We choose to display all responses from -180° to +180°, take the response on axis as reference and choose frequency limits from 150Hz to 15kHz.

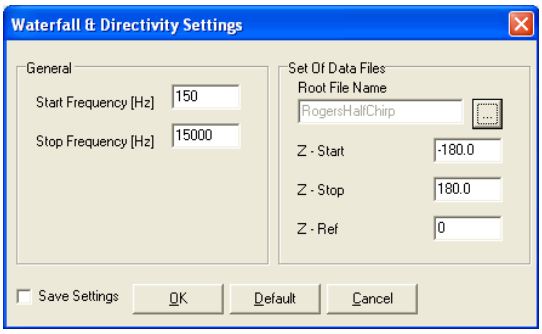


Figure 12.16

We are, at last, ready to start a directivity analysis.

The results are in Fig. 12.17 both as color map and 3-D waterfall plot.

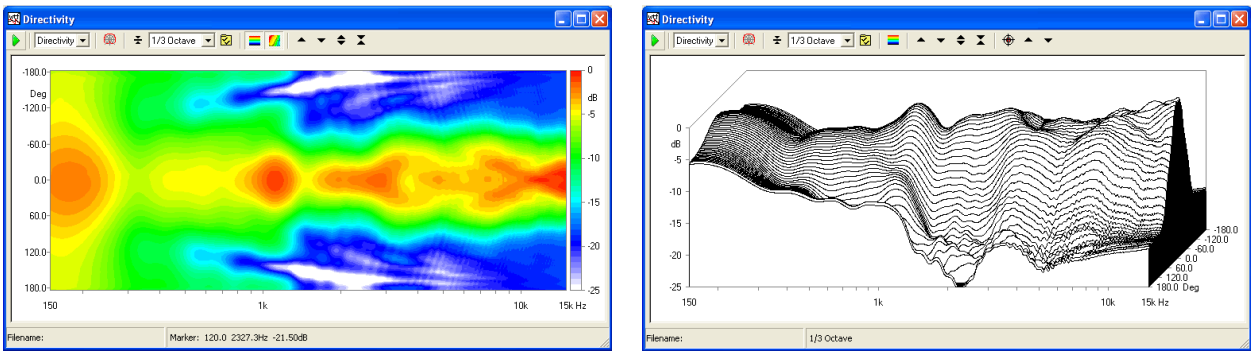


Figure 12.17

Treating polar data we may be more interested in referencing the graph to the response that our speaker shows on-axis. This is easily done pressing the reference button and executing the processing again.

The final result for our polar data waterfall is in Fig.12.18; the response at 0 degrees is now flat and our plot perfectly identifies the behavior of the speaker, providing clear evidence of the different behavior of the polar response versus different frequency zones.

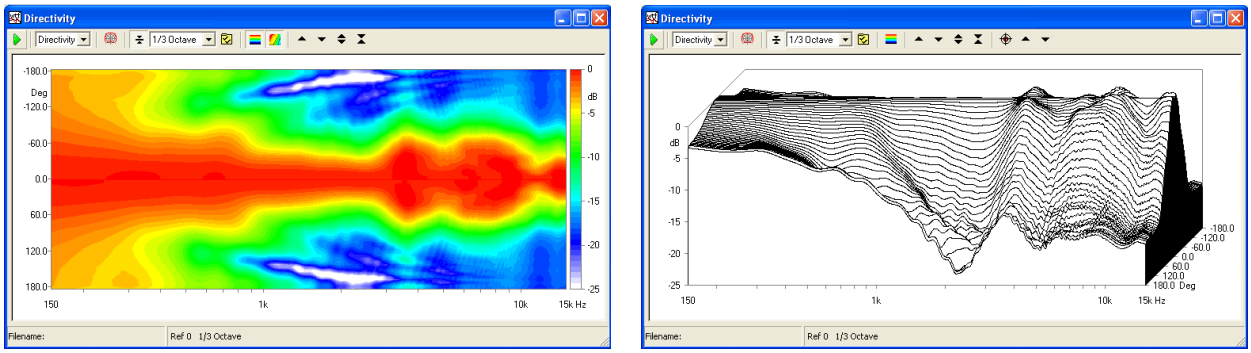


Figure 12.18

Another way to view the same data are the classical circular polar plots. To achieve this ulterior result simply press the Polar Pattern button. Then you may change analysis frequency with the dedicated arrow buttons and save the polar patterns of interest in different overlays reaching a situation like the one in Fig.12.19.

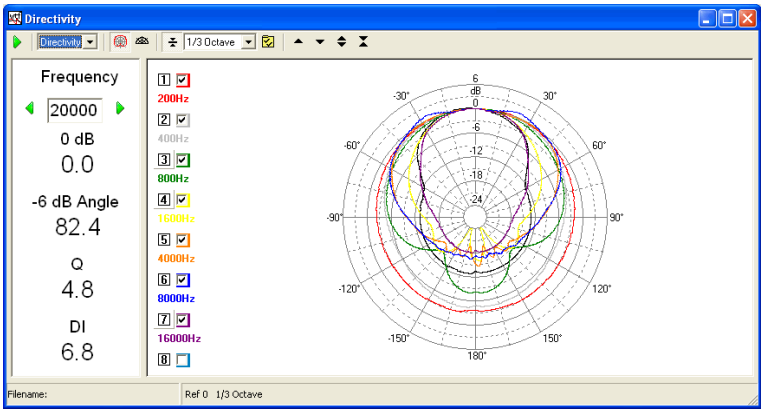


Figure 12.19

12.8 3D SPECIFIC CONTROLS

Please note that the 3D analysis module is available only in the QC version of the software.

Expands the plot changing the balloon radius scale. The balloon radius is reduced.

Compresses the plot changing the balloon radius scale. The balloon radius is increased.

Balloon Top view.

Balloon Bottom view.

Balloon Left view.

Balloon Right view.

Balloon Front view.

Balloon Rear view.

Balloon Perspective view.

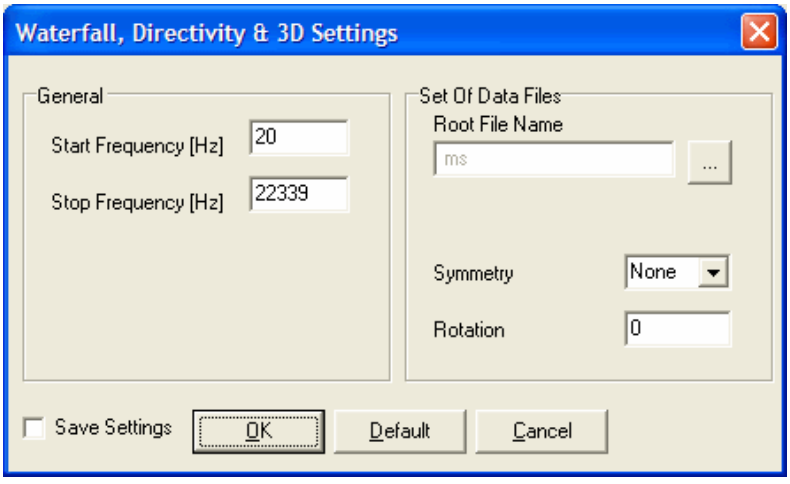
If pressed show the reference balloon.

If pressed the balloon is coloured as function of attenuation relative to the on-axis value.

When in 3D mode moves analysis frequency up 1/3 of octave.

When in 3D mode moves analysis frequency down 1/3 of octave.

12.8.1 3D SETTINGS AND OPERATION



Start Frequency and Stop Frequency

These fields do not have effect on the 3D module. The analysis is always carried out in the 20 Hz to 20 kHz band at 1/3rd octave standard frequencies.

Root File Name and browse button

The name of one file within the set to be displayed. By pressing the associated button it is possible to browse the disk and choose the file.

Symmetry

Select the file set symmetry between: None, Half, Quarter, Axial and H+V.

Rotation

Rotation of the THETA=0 reference angle. The THETA=0 angle is by CLIO conventions oriented as the positive direction of the x-axis. If the data set is saved with a different THETA angle origin the rotation allow to compensate for this. As an example if the data set is saved with THETA origin oriented in the negative direction of the y-axis, the correct setting is Rotation=-90.

The radiation characteristics of a loudspeaker or driver versus frequency and direction rely on a huge amount of data i.e. a set of frequency responses (taken at 5 degree resolution) saved to disk.

With the 3D analysis you get a powerful way for synthesizing a large number of measurements in a single balloon graph.

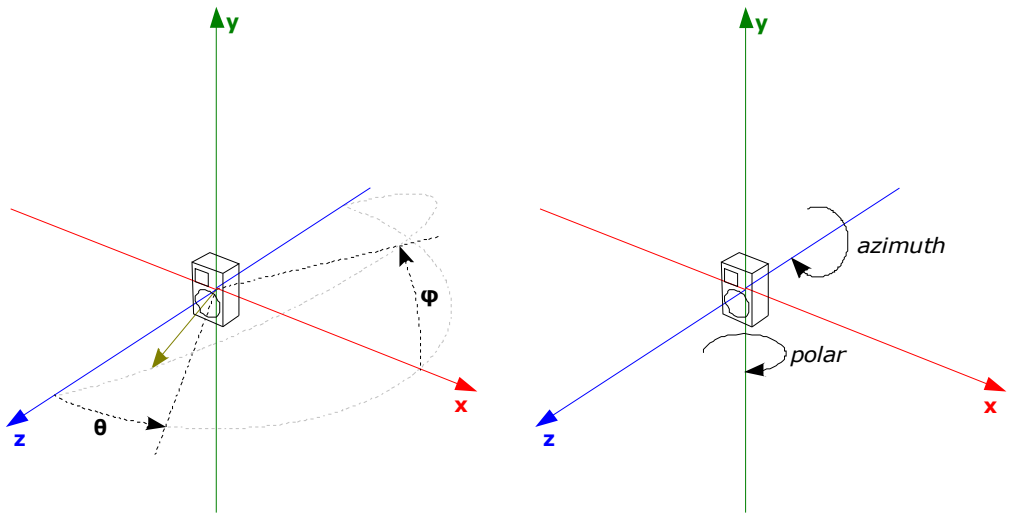
In order to identify the set of files it is important that all of their names follow a particular syntax, that gives certain information to the processing routines.

The syntax follows: **<NAME><PHI*100><THETA*100>.MLS**

NAME is a common file name, PHI is the polar angle and THETA is the azimuth angle. These quantities are separated by **spaces**. While it is possible to give negative numbers to THETA and PHI, we do suggest to collect measurement sets using only positive numbers.

THETA is the polar angle between the loudspeaker axis and the measurement microphone, PHI is the azimuth angle. Positive THETA angles are related to the counterclockwise rotation of the measuring microphone (apparent rotation since it is the loudspeaker that is rotating clockwise) around the loudspeaker. Positive PHI angles are related to the counterclockwise rotation of the DUT around its on-axis direction.

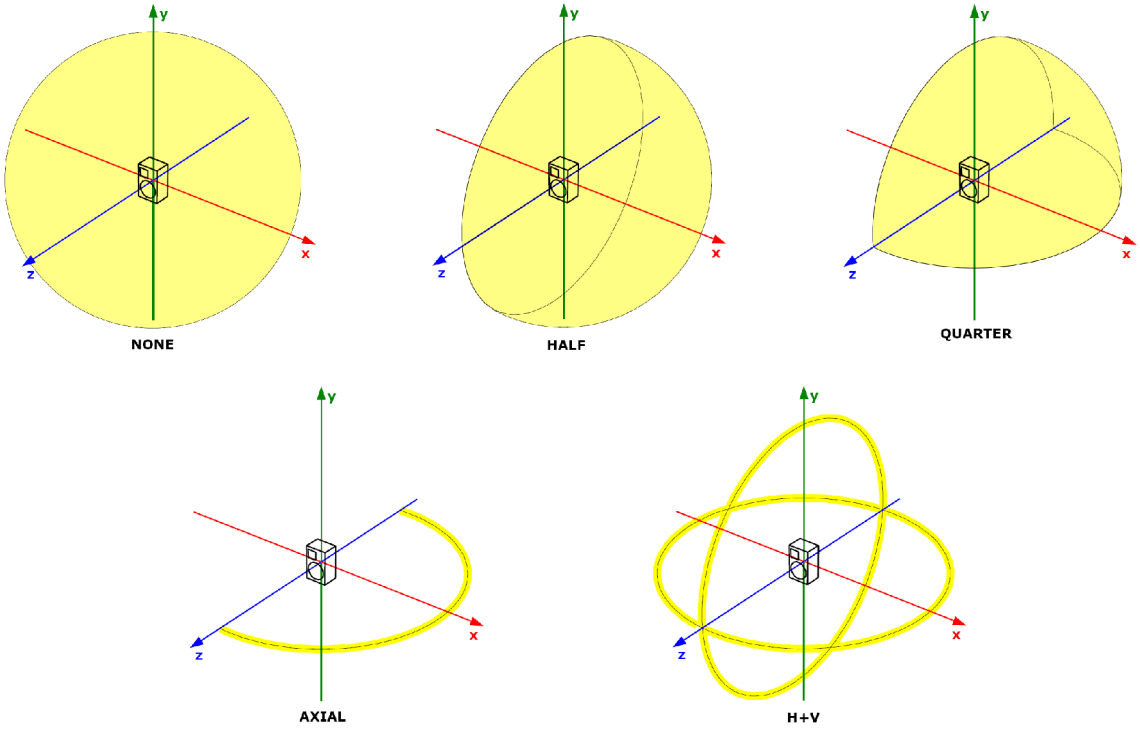
The coordinate system used by CLIO is defined as in the following figures:



The autosaving and naming capabilities of CLIO render the job of measuring and creating a complete 3D directivity data set an easy and automatic task (see later 12.9 and 12.10 for examples).

In order to reduce the number of files needed to describe the directivity pattern of a source, it is possible to use the source **symmetry** if any.

There are five different symmetries available:



The **None**, **Half**, **Quarter** and **Axial** symmetry modes are self explanatory.

The **H+V** mode refer to a slight different scheme and is supported to permit users with a single turntable to create directivity balloons by means of mathematical interpolation of the missing data. The **H+V** mode require a set of 144 measurements, collected over the horizontal and vertical complete polars from -180 to 180 degrees (instead of half polars from 0 to 180 that are used elsewhere).

For a detailed description of **H+V** mode please refer to section 12.13 at the end of this chapter.

12.9 MEASURING LOUDSPEAKER SINGLE POLAR DATA (3D MODE)

Using a single PC controlled turntable (Outline ET2503D or LinearX LT360) under CLIO's control and automation it is possible to easily collect a single loudspeaker polar with a naming convention equal to the 1D Mode (refer to 12.6 and 12.7).

The 3D autosave mode is available with QC version software only.

The advantage of this mode is that the turntable is automatically positioned without any operator intervention.

12.9.1 PREPARING AUTOSAVE AND THE MLS CONTROL PANEL

The MLS measurement should be set in a particular way in order to automatically acquire the responses at various polar angles. **To do this we will use the turntable control and the autosave dialog.**

Let's start with setting the autosave function (see also 5.3). Pressing **Alt-F2** we recall the Autosave Setting dialog (fig.12.20); here we choose the 3D mode and input the desired file name ('RogersHalfChirp'), polar start (-180), polar step (5), polar stop (180), azimuth start (0), azimuth step (0) and azimuth stop (0).

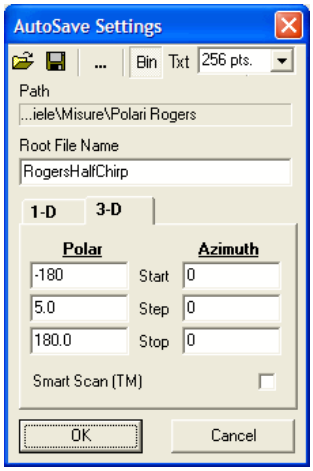


Figure 12.20

By setting the azimuth angles at zero CLIO will save the files as:

<NAME><THETA*100>.MLS

where NAME is a common file name and THETA is the azimuth angle. These quantities are separated by **spaces**, it is possible to give negative numbers to THETA.

The MLS menu is now ready to start taking measurements; we only need to properly set the turntable and its control.

12.9.2 PREPARING THE TURNTABLES

We assume that an ET250-3D turntable is properly connected to your PC (refer to 4.7 for details). To prepare for this measurement session you need to link the turntable. At the beginning of the measurement session or by clicking the sync button on the turntable CLIO will move the turntables to the -180 angle. Thus the turntable must be set up to be in the 0 position with the loudspeaker on-axis with the microphone.

12.9.3 TAKING THE MEASUREMENTS

You are now ready to begin the measuring session. We suggest you to take an initial measurement (with the speaker in place over the turntables in position 0 – on-axis - and with the turntables link button not pressed) to verify all the parameters, especially viewing the acquired impulse response and setting the start and stop values of the measurement window. These values will be applied to all the measurements taken; consider, in this respect, the problem of the trajectory of the acoustic center of the speaker during the rotation.

The last thing to do is to start the procedure by clicking the start button on the turntables control dialog:

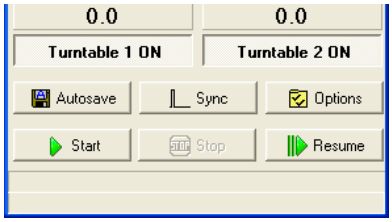


Figure 12.21

A message box alerting about the number of measurements that will be taken is showed:

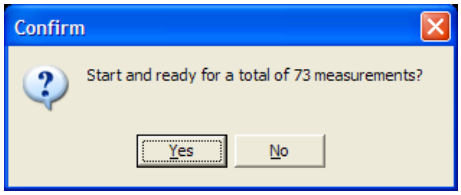


Figure 12.22

After the 73 measurements are taken the session should end while the autosave reset. The procedure can be halted and restarted from the point where it stopped using the stop and resume buttons in the turntables control dialog.

12.10 MEASURING FULL SPHERE LOUDSPEAKER POLAR DATA (3D MODE)

Using one or two PC controlled turntables (Outline ET2503D or LinearX LT360) under CLIO's control and automation is it possible to easily collect sets of loudspeaker (complete or partial) impulse responses balloons.

The 3D autosave mode is available with QC version software only.

The Autosave control panel allows for a simple setup of the measurement set, then CLIO manage the whole process:
- send position commands to the turntables and monitor their status
- perform MLS measurements and save the results with the naming rules

12.10.1 PREPARING AUTOSAVE AND THE MLS CONTROL PANEL

We will use two PC controlled turntables (Outline ET250-3D) under CLIO's control to gather the full sphere balloon response of a loudspeaker.

The MLS measurement should be set in a particular way in order to automatically acquire the responses at various polar and azimuth angles. **To do this we will use the turntable control and the autosave dialog.**

Let's start with setting the autosave function (see also 5.3). Pressing **Alt-F2** we recall the Autosave Setting dialog (fig.12.23); here we choose the 3D mode and input the desired file name ('RogersHalfChirp'), polar start (0), polar step (5), polar stop (180), azimuth start (0), azimuth step (5) and azimuth stop (355).

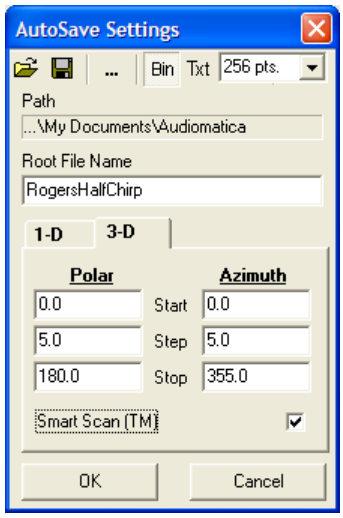


Figure 12.23

The MLS menu is now ready to start taking measurements; we only need to properly set the turntable and its control.

12.10.2 PREPARING THE TURNTABLES

We assume that the ET250-3D turntables are properly connected to your PC (refer to 4.7 for details). To prepare for this measurement session you need to link the turntables. At the beginning of the measurement session or by clicking the sync button on the turntable CLIO will move the turntables to the 0,0 angle. Thus the turntables must be set up to be in the 0,0 position with the loudspeaker on-axis with the microphone.

12.10.3 TAKING THE MEASUREMENTS

You are now ready to begin the measuring session. We suggest you to take an initial measurement (with the speaker in place over the turntables and with the turntables link button not pressed) to verify all the parameters, especially viewing the acquired impulse response and setting the start and stop values of the measurement window. These values will be applied to all the measurements taken; consider, in this respect, the problem of the trajectory of the acoustic center of the speaker during the rotation.

The last thing to do is to start the procedure by clicking the start button on the turntables control dialog:

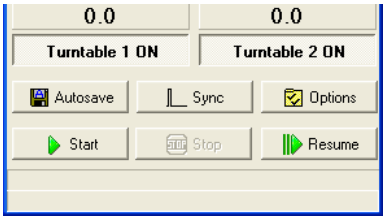
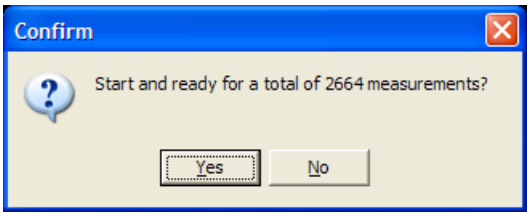


Figure 12.24

A message box alerting about the number of measurements that will be taken is showed:



Press Yes. After each MLS measurement is taken you will see the turntables rotating and CLIO waiting the specified delay time before automatically taking the next measurement. Should this time be insufficient you have to reset the turntable delay value accordingly. The autosave function will refresh the file name after each measurement (Fig.12.25).

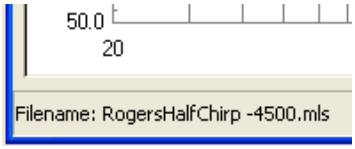


Figure 12.25

After the 2664 measurements are taken the session should end while the autosave reset. The procedure can be halted and restarted from the point where it stopped using the stop and resume buttons in the turntables control dialog.

12.11 REPRESENTING 3D BALLOON DATA

To represent and export the measured 3D directivity data we need to select the 3D mode in the Waterfall, Directivity & 3D control panel (**The 3D analysis mode is available with QC version software only)**. Then enter the 3D settings dialog and press the browse button. Entering our data directory we find the situation in Fig.12.26:

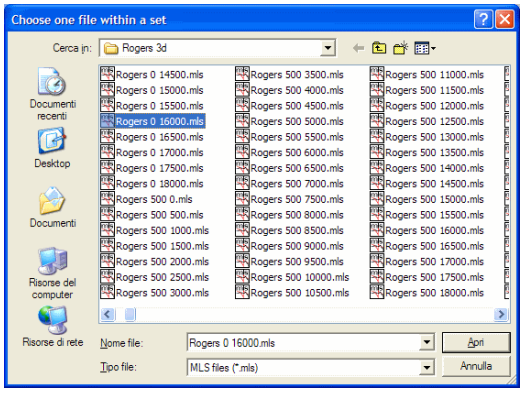


Figure 12.26

The set of files is composed by a certain number of files; it is sufficient to choose one of them. It is now important to set the symmetry mode and the rotation angle for the data set, this is done using the combo box for the symmetry and editing the rotation angle as in Fig.12.27. We choose None for the symmetry and 0 for the rotation, since we are analyzing a complete full sphere measured set.

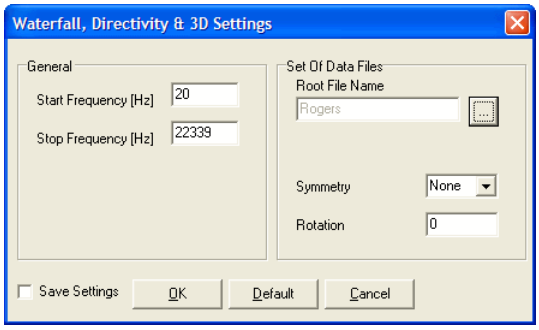


Figure 12.27

We are, at last, ready to start a 3D analysis. The result is in Fig. 12.28 as color balloon at 1 kHz.

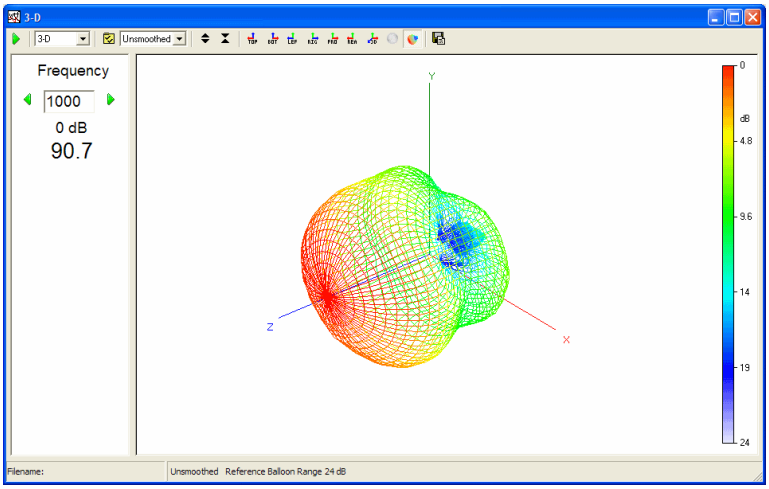


Figure 12.28

It is possible to inspect the 3D directivity of the source selecting a 1/3rd octave band from 20 Hz to 20 kHz and rotating the balloon view. To rotate the balloon it is possible to select one of the predefined views by pressing the view buttons, or click and drag the balloon.

Figure 12.29 show the balloon response at 5 kHz.

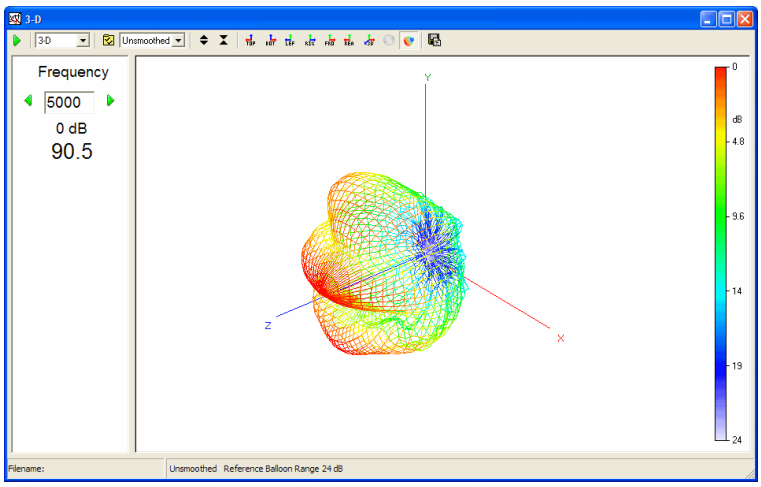


Figure 12.29

Figure 12.30 and 12.31 are showing different views (top and right) of the same balloon response at 3150 Hz.

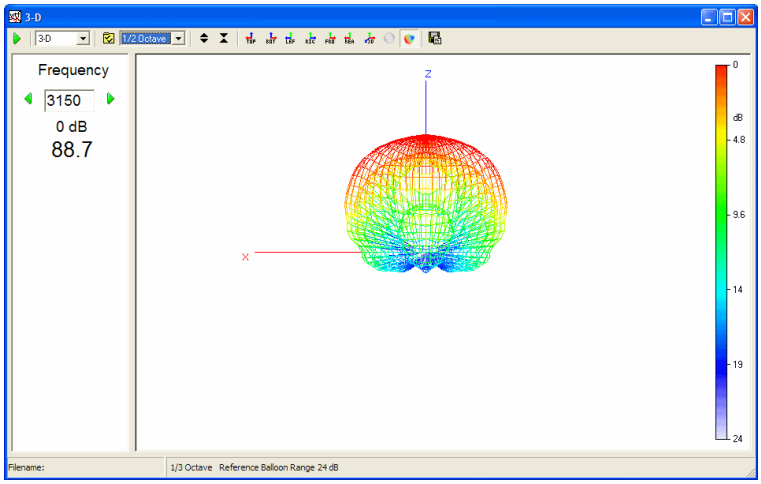


Figure 12.30

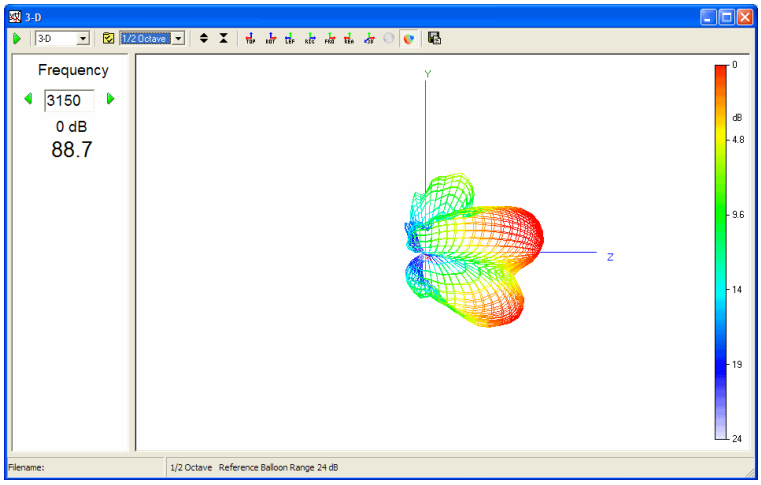


Figure 12.31

12.12 EXPORT 3D BALLOON DATA

The 3D mode feature a powerful tool to export the measured data towards the most common simulation software formats. The supported export format are:

EASE .xhn
EASE .xhn ASCII format (only module, no complex data).

CLF v2 .tab
Common Loudspeaker Format CLF v2 .tab ASCII format.

Impulse Responses
Set of Impulse Responses in ASCII .txt format ready to be imported with EASE SpeakerLab.

The process of creation of the loudspeaker model for a simulation software require the measurement of the directional response of the loudspeaker. Please check 12.10 for more information.

Once the 3D data set is measured, with the 3D analysis tool it is possible to check data consistency and inspect the directional characteristics of a source.

The Export Balloon button opens the Balloon Export form.

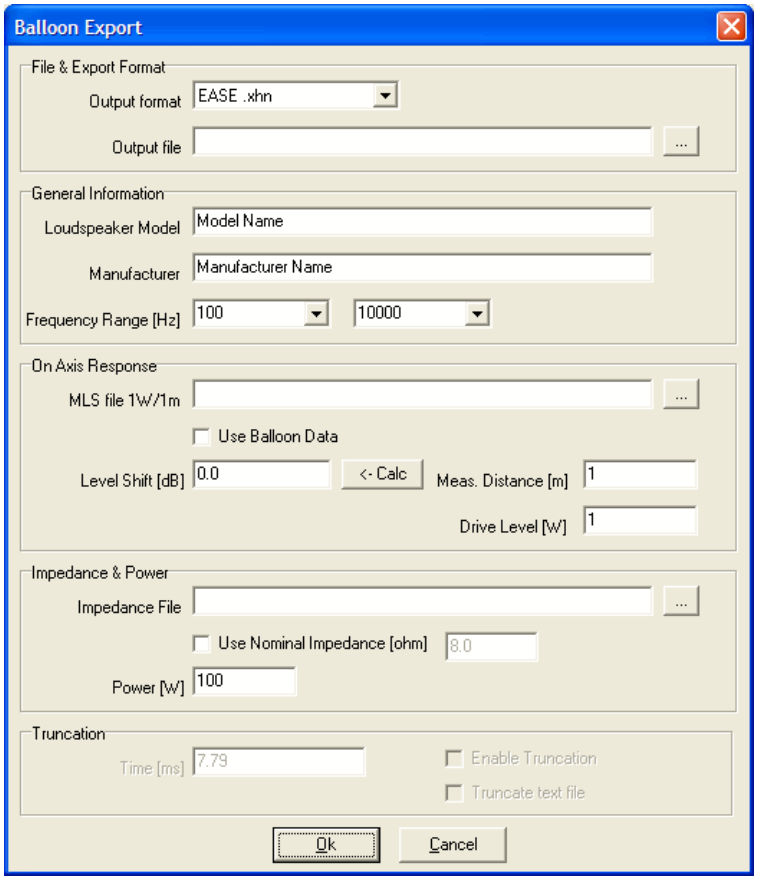


figure 12.32

12.12.1 EXPORT EASE .XHN AND CLF V2 .TAB FILES

In case of **EASE .xhn** and **CLF v2 .tab** format is selected in the File And Export Format Group then the General Information, On Axis Response and Impedance & Power groups are active.

Output File defines the file name and location where the file will be saved; it is possible to choose it clicking on the browse for Choose Output File button (...).

The **Loudspeaker Name** and **Manufacturer Name** fields will be used into the exported text file.

The **Frequency Range** can go from 100 Hz to 10 kHz in case of EASE .xhn, from 25 Hz to 20 kHz in case of CLF v2 .tab. The frequency range to export can be modified using the combo boxes.

The On-Axis Response group let the user decide to use the data set on-axis measurement or use an MLS file as on-axis reference. It is also possible to apply a **Level Shift** to our measured response to correct for power and distance different from the required 1W/1m. A basic calculation that calculates the required level shift as function of the **Drive Level** in W applied and **Measurement Distance** between loudspeaker and microphone.

Similarly to the previous group, the Impedance & Power group let the user decide to use a real measured impedance in .sini format or a nominal value. Into the same group the user can specify a power level (valid for all bands) for the source.

It is clear that the functions of this export menu are minimal, but they successfully meet the user need to create loudspeaker models used by the most common simulation software.

Once the output text file is saved, it may be necessary to preprocess with a text editor to add or modify information.

Figure 12.33 report an example of a data set exported to CLF v2. tab format and imported by the CLF software.

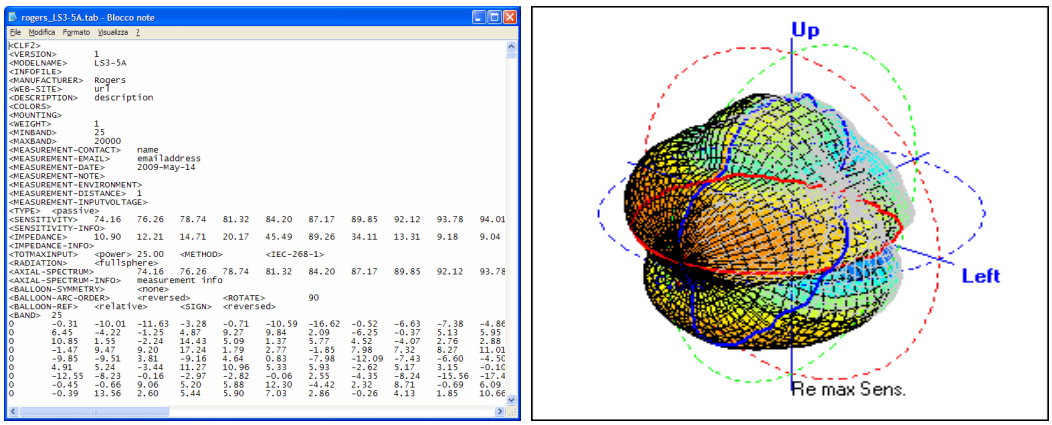


figure 12.33

12.12.2 EXPORT SET OF IMPULSE RESPONSES

If **Impulse Response** is selected only the last Truncation group is active.

The **Output Folder** define the path where the Impulse Responses in text format will be saved. The file will be saved as Time Data impulse responses with the naming convention requested by the EASE SpeakerLab:

IR <PHI*100> <THETA*100>.txt

If the **Enable Truncation** option is selected the time response is windowed with a rectangular window with **Time (ms)** duration.

If the **Truncate Text File** option is selected the exported text files are limited to the points inside the time window. This option reduce drastically the size of the exported data sets.

12.13 H+V MODE

In this mode the 3D analysis module requires only 2 sets of polar measurements with 5 degrees resolution: one for the horizontal plane and one for the vertical plane.

This reduced data measurement set is then mathematically interpolated in order to create the full sphere balloon.

The data set requested require the full horizontal polar from -180 to 180 degrees and the vertical polar from -180 to 180 degrees.

The naming should look like:

<NAME> 0 <THETA*100>.MLS

for the horizontal polar and:

<NAME> 9000 <THETA*100>.MLS

for the vertical polar.

To collect the H+V data there are two possible approaches, using autosave in 1D mode or 3D mode.

In case of the simpler 1D mode, when collecting the horizontal polar, the root filename must include the ' 0' at its end. For the vertical polar the rootfile name must include the ' 9000' at its end. The loudspeaker must be manually positioned in horizontal and vertical positions.

If computer controlled turntables are available it is also possible to use the 3D autosave mode. In this case it is necessary only to set the start, stop and step values. Please use these settings:

Polar start -180: stop 180: step 5 degrees

Azimuth start 0; stop 90; step 90 degrees.

13 MEASURING IMPEDANCE AND T&S PARAMETERS

13.1 INTRODUCTION

This chapter deals with impedance measurements generally before going onto the Thiele and Small Parameters Menu description. CLIO performs impedance vs. frequency measurements both from within the MLS and the Sinusoidal Menu. You will find specific information in the relative Chapters. Both are relevant to what we will now explain. Here we explain connections, principles and other topics that apply to both menus. Differences and choice criteria are also covered.

13.2 GENERALS

Whatever the stimuli, CLIO sees a voltage at its input. Impedance is obtained by a suitable post process, which changes depending upon how the user decides to perform the measurement. Four methods are available. Two of them, **Internal** and **I Sense**, were already found in the Settings Dialog, both in MLS and Sinusoidal. We will start with these, leaving **Constant Voltage** and **Constant Current** to later on. The last two methods derive their name from the description of what is going on during the measurement; they were the standard before automatic computer based Measuring Systems. **Internal** and **I Sense** are faster, requires only one measurement, and handles the connections needed much more easily.

13.3 INTERNAL MODE

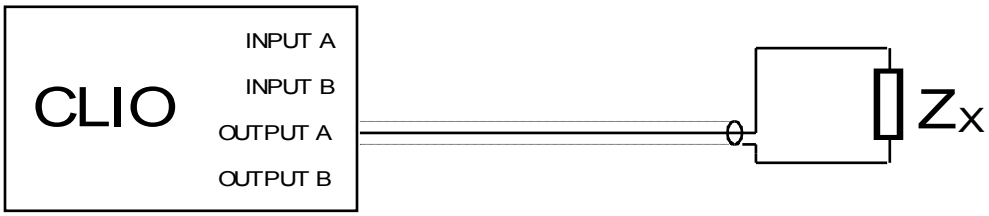
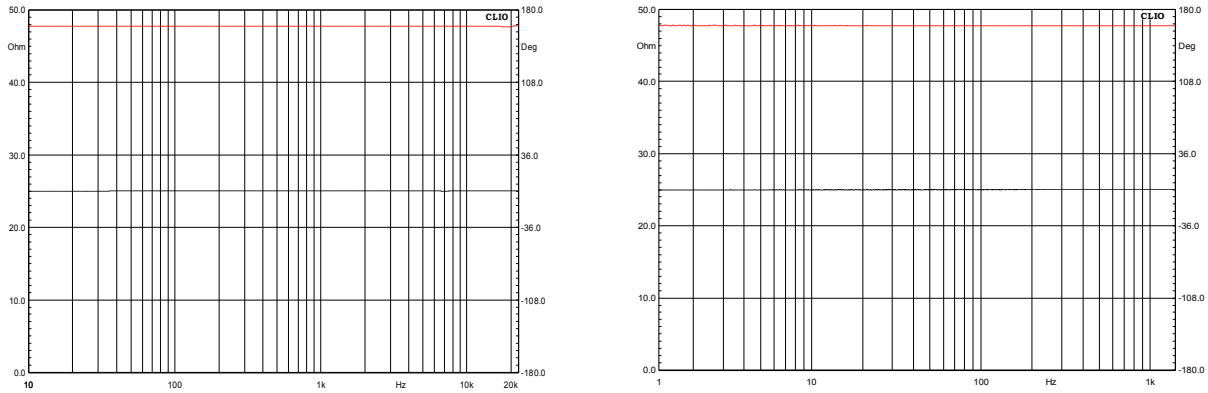


Figure 13.1

In principle, internal mode is very similar to **Constant Current**. It takes advantage of the knowledge CLIO has of its own output level and output impedance (660 Ohm 0.1%). In fact, this value is lower than what should be used for a true Constant Current procedure. Veterans should not worry: even if the resistor value is on the same order of magnitude as the device you are going to measure instead of 10 times higher as usually advised, no errors occurs as CLIO is able to acquire and evaluate the signal as a complex signal (real and imaginary) at the device terminals. Fig.13.1 shows the connections for measuring Impedance in Internal mode. Only one cable (XLR-to-alligators like the one in the photo) is needed, without any other external active or passive devices. Before starting any impedance measurement in Internal Mode remember **to switch on the loop button** (or) of the channel you are going to use.

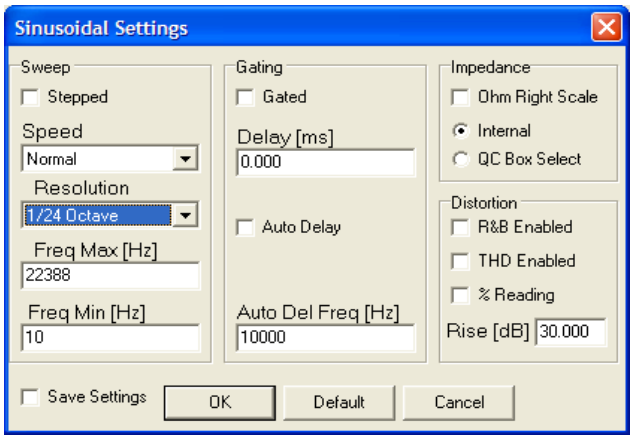
If you are a novice in using CLIO, or to impedance measurements in general, use this mode; also do not start measuring loudspeaker impedance immediately. Get a 22 to 100 Ohm resistor, possibly 1% tolerance, and gain experience with something which you should already know the expected results of. Here are two examples both with Sinusoidal and MLS. Before you press go, remember to set the Y scale to Ohm. For this example we chose a 47 Ohm resistor. The modulus should be very close to the resistor value (red curve) and equally important, the shown phase should be very close to 0; these results should cover the entire frequency range. Notice that Fig.13.3, which refers to MLS, extends low frequency limit to 1Hz while Sinusoidal, Fig.13.2, stops at 10Hz which is the lowest possible. If you are not getting similar results do not proceed with a loudspeaker impedance measurement. To do so would only add problems later by using an inaccurate working procedure, as we will soon see. Even in internal mode CLIO's output level is left to the user. We set it at +10dBu here as resistors are, hopefully, the most linear devices we can get. Since we had very comfortable results lets proceed with a Loudspeaker.



Figures 13.2 and 13.3

13.3.1 MEASURING IMPEDANCE OF LOUDSPEAKERS

We will start with a 5" woofer using Sinusoidal, **our preferred choice**, with the following Settings (1/24 of octave resolution).



Besides frequency range, which can be changed without side effects, those above are problem free settings for impedance measurements. We will experiment a little, pointing out difficulties that might arise. Let's start with output level, which is a sensitive topic.

13.3.2 SETTING THE RIGHT LEVEL

The five curves of Fig.13.4 are taken at 5 different output levels, ranging from 10dBu to -10dBu in 5dB steps. The red curve refers to +10dBu, the blue to +5dBu, the remaining are substantially overlapped.

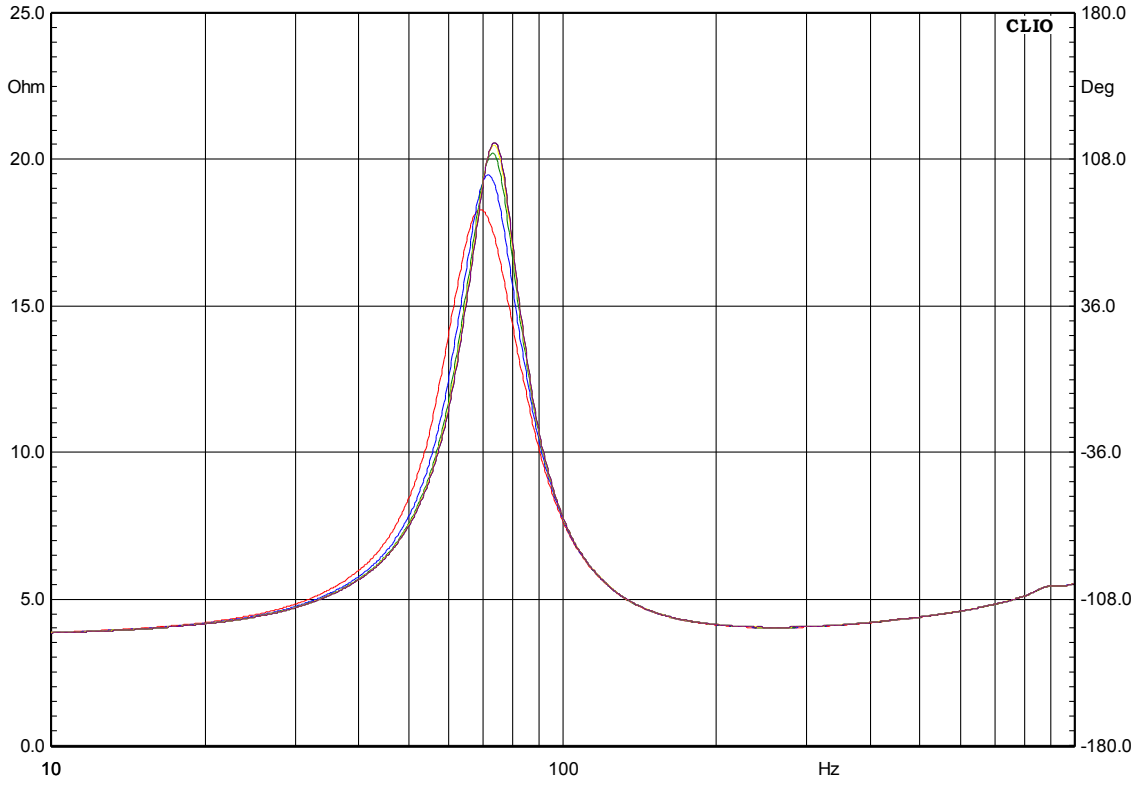


Figure 13.4

It turns out that going from a linear device such as our resistor to loudspeakers

makes life harder even if more interesting. Deriving the principals T&S Parameters from the five curves yields to Table 13.1

	Fs	Qms	Qes	Qts
+10dBu	69.244	3.105	0.609	0.5094
+5dBu	71.63	3.6461	0.6643	0.5619
0dBu	72.9912	3.986	0.695	0.5920
-5dBu	73.5429	4.1663	0.7147	.61
-10dBu	73.82	4.227	0.7218	0.6166

Table 13.1

Values from 0dBu to -10dBu are in optimum agreement and this sets the maximum level to be used to 0dBu. Interestingly enough, **Internal Mode** is less sensitive to output level. We will go further into this topic relying on CLIO being a complete and powerful audio measuring system. Without changing connections we will use CLIO's FFT and Signal Generator to evaluate the distortion current at resonance at 10dBu. Fig.13.5 shows the spectrum in this condition. The second harmonic is 56dB lower than the fundamental, which is 0.158% distortion. Even at 10dBu we are easily in the linear region of the loudspeaker motor. However what we have seen above, clearly states 10dBu is quite a high level for this device.

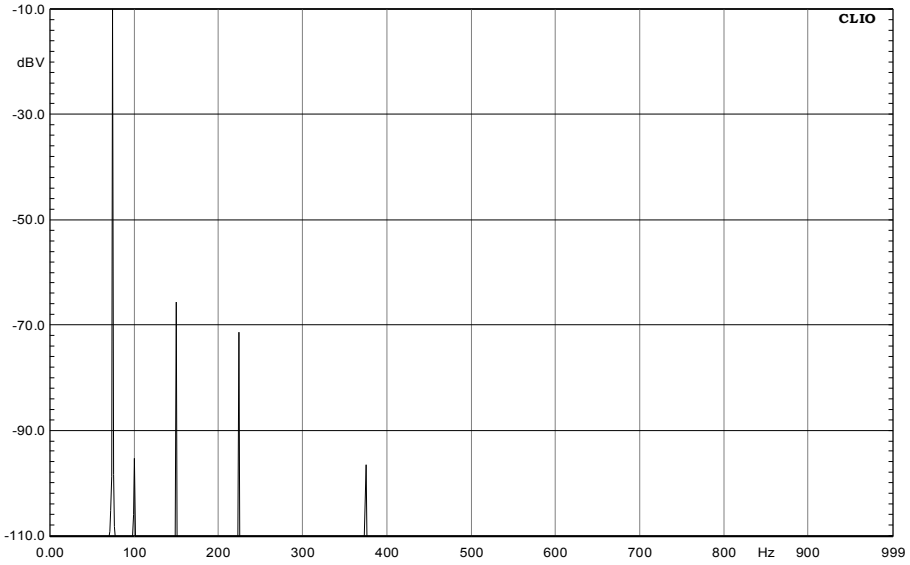


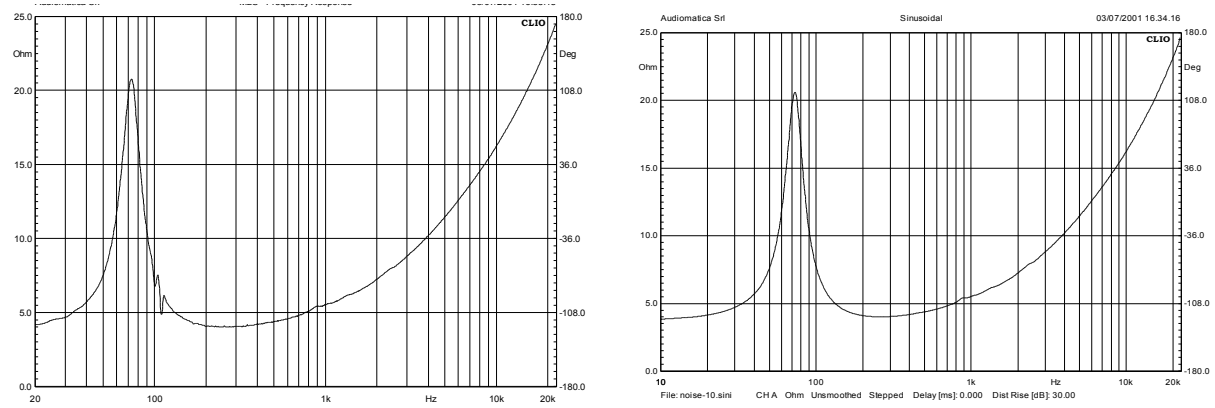
Figure 13.5

The reader could be tempted to determine the absolute quality of the device by means of this procedure. While he or she may be right, a lot of experience is needed. One important parameter, for reliable comparison among devices, is the value, in VRMS, at the speaker terminals at resonance. It is shown from FFT as -10dBV that is 0.316VRMS.

13.3.3 DEALING WITH ENVIRONMENTAL NOISE

The next problem in measuring loudspeaker impedance is noise. Transducers do their job in both directions and noise will appear as voltage exactly where CLIO's input is connected. To evaluate the problem we deliberately produced a disturbance by generating a 110Hz single tone causing 58 dBSPL at the speaker cone. We took two impedance curves in this condition one with MLS the second with Sinusoidal.

Both were taken at-10dBu, a value that gained our favor before. Results are in Fig.13.6 for MLS and Fig .13.7 for Sinusoidal.



Figures 13.6 and 13.7

This is one reason why we prefer sinusoidal analysis to measure impedance.

13.3.4 DEALING WITH VIBRATIONS

The last enemy we must consider is external vibrations.

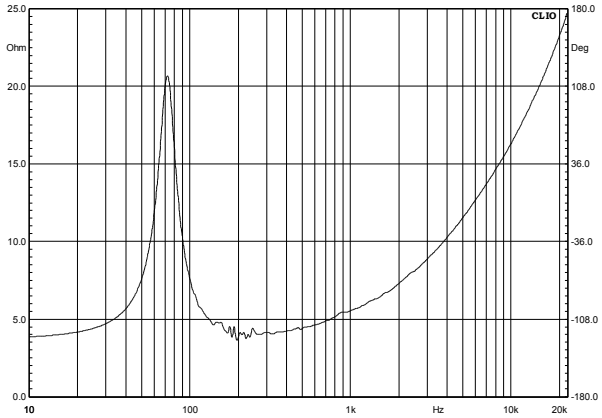


Figure 13.8

Fig.13.8 is an impedance curve taken with the loudspeaker positioned on a computer table, on one of those appendages that may be set up or down upon needs. This support clearly resonates, exited by the loudspeaker, at around 200Hz. No matter how good the loudspeaker is fixed to a structure if the structure itself moves then potential problems may become apparent. Usually there is no need to fix anything as long as you are using a stable, non-resonating structure.

Up to now we have dealt with Internal Mode. We will briefly go through the others modes; problems found up to here will remain the same or worsen from here on!

13.4 I SENSE

This requires Audiomatica CLIOQC Amplifier and Switch Box model 2, 3 4 or 5. It is a simplified Constant Voltage method. Simplification arises as both device gain and sensing resistor (around 0.1 Ohm) is known. Fig.13.9 shows the CLIOQC Software Control Dialog Box. I Sense should be selected.

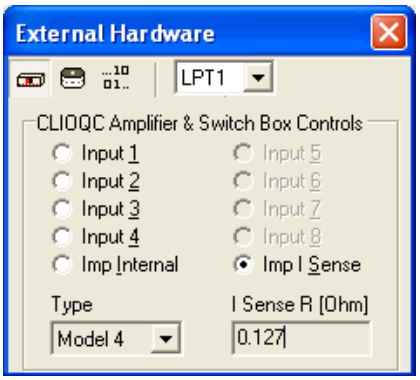


Figure 13.9

Fig.13.10 shows required connections.

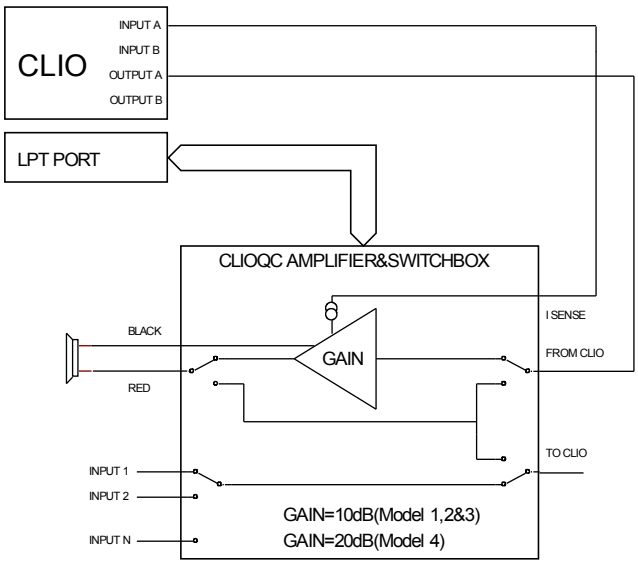


Figure13.10

Before proceeding, whether in MLS or Sinusoidal, remember to go into the Settings Dialog and select I Sense under Impedance. CLIO's output level has to be set to a much lower level than in Internal Mode. We have seen before that with 10dBu out we had 0.316V at the speaker terminals at resonance. Should you keep this level you would have 7.75V at any Frequency! -20 to - 30 dBu are reasonable output levels to start with. The CLIOQC Amplifier gain is 10dB (Model 2&3), 20dB (Model 4) or 26dB (Model 5); it therefore multiplies CLIO's output by 3.16 (Model 2&3), 10 (Model 4) or 20 (Model 5). As the output impedance is close to 0 no further attenuation occurs once the speaker is connected. Finally we shall deal with measuring accuracy in this case. The sensing resistor value has been stated as around 0.1 Ohm. It is very difficult to keep such a low value within a reasonable tolerance and therefore the real value is going to change between units. The default value that is used is 0.127 Ohm. The user can further adjust this value using a reference resistor of known value in the 10 Ohm range (the real value can be eventually measured with high accuracy using the internal mode). Simply take an impedance measurement of the known resistor and adjust the I Sense value

displayed multiplying it by the ratio between the known resistor value and the marker reading at 1kHz. For example: assume a known resistor value 10 Ohm, reading at 1kHz 9.3 ohm and an I Sense value of 0.127 Ohm. Multiply 0.127 by 1.075268817 to obtain 0.13655914, input this new value and check everything by performing a new measurement.

13.5 CONSTANT VOLTAGE & CONSTANT CURRENT

These were the standard approaches to measuring impedance with a traditional set of instruments. We will skip further theoretical discussion and go directly on how CLIO implements them. Both these methods require two external components, a reference resistor of suitable and known value (where known means better than 1% tolerance) and a power amplifier. They also require two measurements to be taken one after the other, changing connections in between. CLIO, by means of its processing tools can speed things up a lot but the whole procedure remains quite complicated. Whatever you are going to use, MLS or Sinusoidal, all the measurements should be performed in Volts (the Y Scale should be dBV).

13.5.1 CONSTANT VOLTAGE

Proceeding step by step we are going to create two files, one named "reference", the second named "device". The two measurement files must have identical settings and identical CLIO's output level. We choose a sensing resistor, Rs, of 1 ohm at 1% tolerance. Fig.13.11 shows connections for creating the reference file. The level you choose now is the total measuring level.

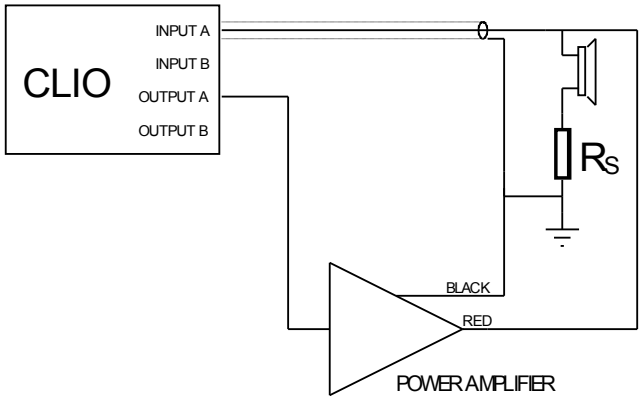


Figure 13.11

The reference measurement is shown in Fig 13.12. As expected the output of the amplifier appears as a straight line. Should this not be so, this would not be a problem as the whole procedure compensates for any deviation in the frequency response. More interesting is the absolute level. As the Rs value is small, this level appears nearly unchanged to the speaker. We read -12dBV that translate in 250mV.

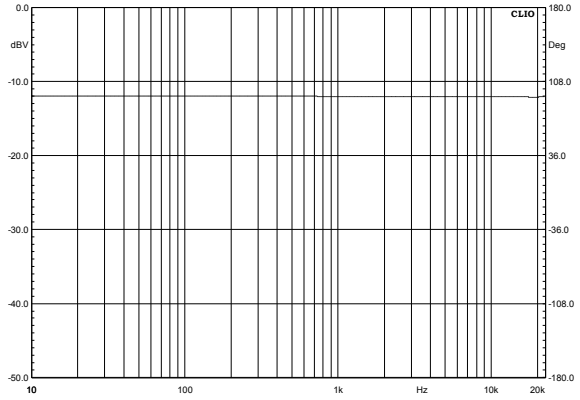


Figure 13.12

Let's now proceed with measuring the device. Connections need to be changed as in Fig. 13.13. We are now going to measure the voltage across Rs, which is proportional to the current in the device. Leaving everything in the Sinusoidal menu as it was for the previous measurement we obtain Fig.13.14. It seams a strange shape if you are not used to. But in fact we are measuring Current here and, Voltage being constant, it decreases at resonance (from Ohm's law I=V/R).

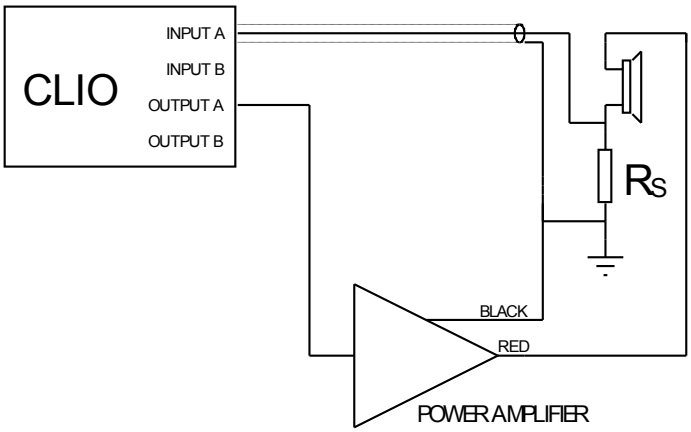


Figure 13.11

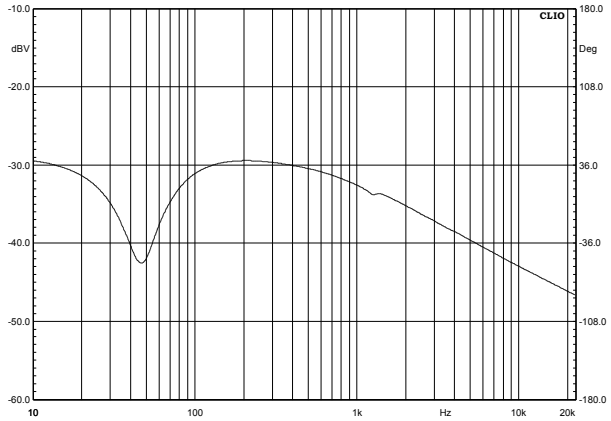
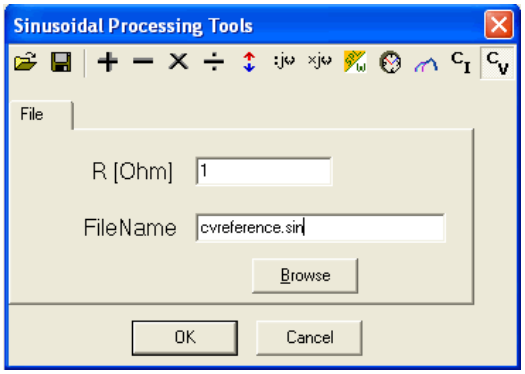


Figure 13.14

What we do from now on is just post processing. Leaving the last measurement in memory, open the Processing Tools Dialog from within Sinusoidal (MLS has the same), select the CV icon and input the data you should know.

The Ohm value is that of the Sensing Resistor while in the edit box you see the name of our previous file, cvreference.sin, including its own path which could be different in your case.



Pressing OK we get Fig.13.15 which is our final result. Note that the Y Units have been changed to Ohm. This result is only in memory and should be saved now for further use.

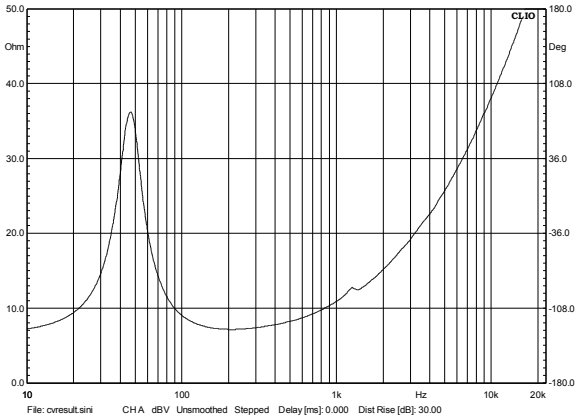


Figure 13.15

13.5.2 CONSTANT CURRENT

We will go quicker now as it's very similar to what we have just seen. Connections for creating the reference are the same, please refer to Fig.13.11. Again everything should be left unchanged between creating the reference and the device files. The big difference from before is the output level. Here we choose a 1kOhm resistor. This will attenuate a lot of the signal at the loudspeaker terminals. Fig.13.16 shows the reference measurement.

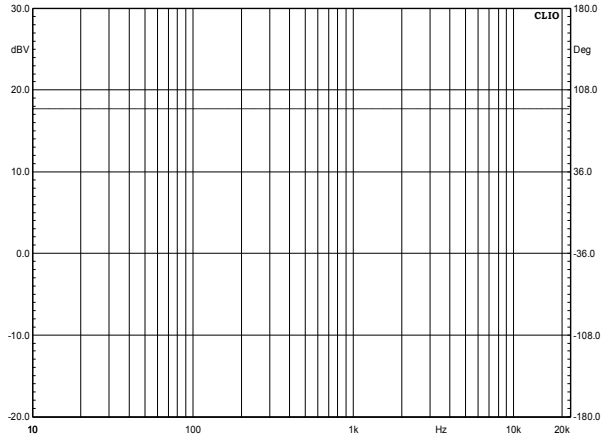


Figure 13.16

We have a straight line again; notice the level has became 17dBV now. We change connections, in accordance with Fig.13.17, and nothing else.

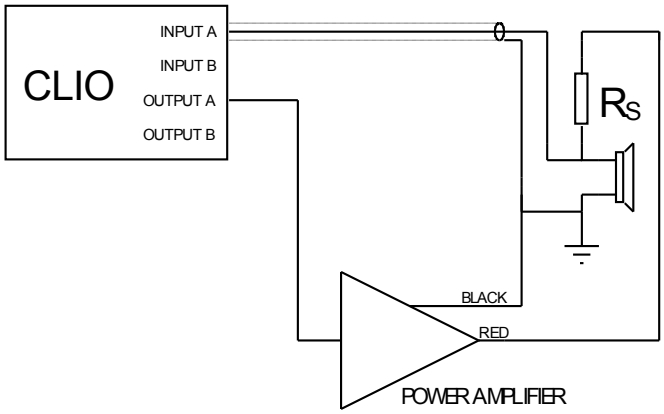


Figure 13.17

The figure shows us we are going to measure the voltage across the device. Therefore the next graph, Fig.13.18, will give us detailed information regarding the measuring level.

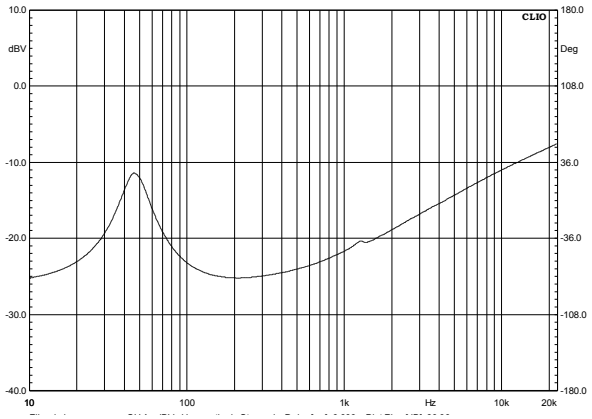
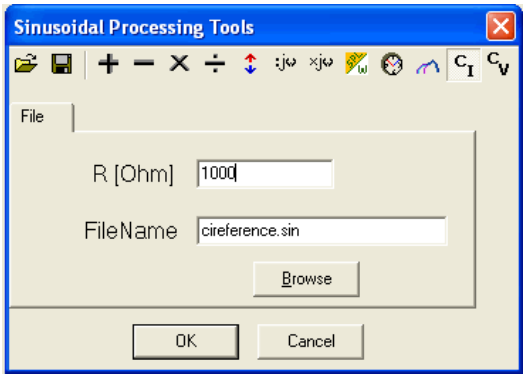


Figure 13.18

The level is low enough not to worry us and we proceed with our post process as before. Instead of the CV button CI should be selected.



Finally Fig.13.19 shows the process result.

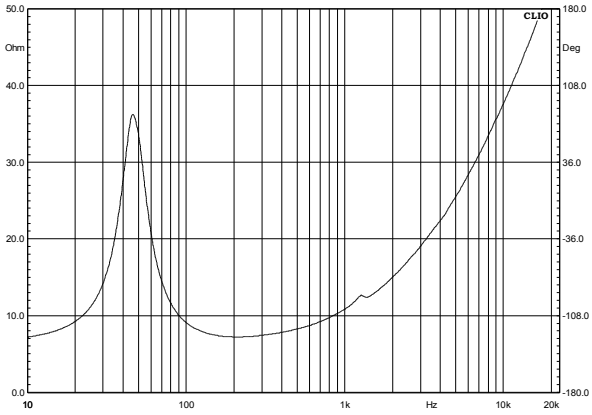


Figure 13.19

13.6 IMPEDANCE: SINUSOIDAL OR MLS

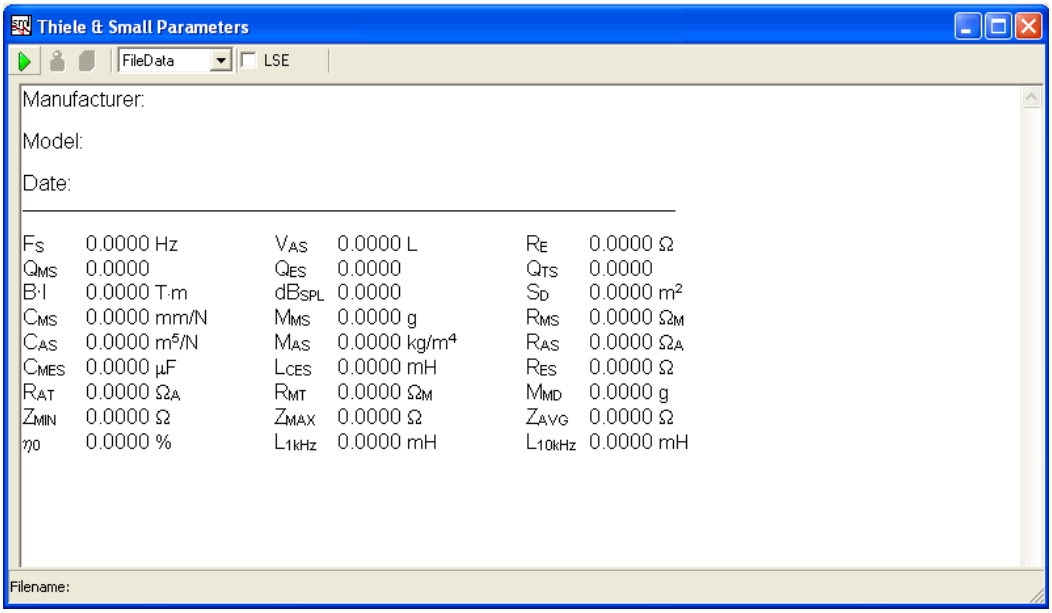
Up to now we have almost always used Sinusoidal to perform Impedance Measurements. When MLS has been used, it was to point out problems. We also stated Sinusoidal is the preferred choice. The inexperienced user might ask why the MLS Menu has impedance measuring capability at all. In fact MLS yields accurate Impedance Measurements. It is, however, less likely to forgive errors and/or measuring set-up problems of any kind. Users, however, should always take the maximum effort to avoid these situations anyway. Furthermore MLS has several advantages over Sinusoidal. The most evident is execution time which is significantly better, even using big sizes, which, by the way, is always advisable. It can measure down to 1Hz while Sinusoidal stops at 10Hz. Its high sensitivity to even weak non-linearity can be used to reveal even small rub & buzz problems, directly from the impedance measure. We advise you to use Sinusoidal measurement techniques initially until you become more experienced with CLIO. After which everyone will be able to form an opinion based on personal experience.

13.7 THIELE & SMALL PARAMETERS

13.7.1 INTRODUCTION

CLIO handles Thiele and Small Parameters, hereafter referred to as T&S, as a post process of impedance measurements. Three options are available for source data, selected by the Data Origin Drop Down Control: Sinusoidal Impedance Data, MLS Impedance Data, File Data, the last created with either of the previous. There are no conceptual differences between File and the first two options, beside where the data resides. Choosing Sinusoidal or MLS, the user should have one of these measurements in memory while the File Data option will prompt for file selection. Impedance measurements are therefore the central topic and the previous paragraphs are an essential background. To get the full set of T&S, two impedance curves are needed. The first is the loudspeaker's free air impedance, the second, the impedance obtained either with Delta Mass or Delta Compliance method. The first consists in adding a suitable Known Mass to the Loudspeaker cone, the latter in loading the cone with a box of Known Volume.

13.7.2 T&S PARAMETERS CONTROL PANEL



The User Interface is quite simple with three Buttons, one Drop Down and one Check Box.

Starts a T&S Parameters procedure. See 13.7.3 for details.

Delta Mass T&S Parameters calculation. See 13.7.3 for details.

Delta Compliance T&S Parameters calculation. See 13.7.3 for details.

The Drop Down allows selecting the data origin as described in the introduction. The LSE Check Box, when checked, enables a Least Square Error routine while calculating T&S Parameters; more on this later. Besides information regarding the device tested, the Control Panel displays 27 parameters. Here is what they are.

13.7.3 GLOSSARY OF SYMBOLS

F_S	Resonant frequency of driver including air load
V_{AS}	Volume of air having the same acoustic compliance as driver suspension
R_E	DC electrical resistance of voice coil
Q_{MS}	Q of driver at Fs considering mechanical losses only
Q_{ES}	Q of driver at Fs considering electrical resistance only
Q_{TS}	Q of driver at Fs considering all driver losses
$B \cdot l$	Motor strength, product of B times l
dB_{SPL}	Acoustic pressure produced by the driver at 1m when driven driven by 2.83 V
S_D	Effective surface area of the driver cone
C_{MS}	Mechanical compliance of driver suspension
M_{MS}	Mechanical mass of driver cone assembly including air load
R_{MS}	Mechanical resistance of driver suspension losses
C_{AS}	Acoustic compliance of driver suspension
M_{AS}	Acoustic mass of driver cone assembly including reactive air load
R_{AS}	Acoustic resistance of driver suspension losses
C_{MES}	Electrical capacitance representing the driver total moving mass
L_{CES}	Electrical inductance representing the driver mechanical compliance
R_{ES}	Electrical resistance representing the driver mechanical losses
R_{AT}	Total acoustic resistance of driver
R_{MT}	Total mechanical resistance of driver (suspension losses + electrical reflected)
M_{MD}	Mechanical mass of driver cone assembly excluding air load
Z_{MIN}	Minimum impedance in the frequency range above Fs
Z_{MAX}	Impedance at Fs
Z_{AVG}	Average of impedance modulus over the measured frequency limit
$s\eta_0$	Efficiency
L_{1kHz}	Inductance at 1kHz
L_{10kHz}	Inductance at 10kHz

13.7.4 T&S STEP BY STEP

Getting T&S requires two impedance measurements. As we will use both methods we need three, the first relative to the driver in free air, the second to the driver with a known mass (Delta Mass) added to the cone, the third to the driver loaded with a known volume (Delta Compliance). Fig.13.20 shows the results of the three measurements, overlaid in one single graphic.

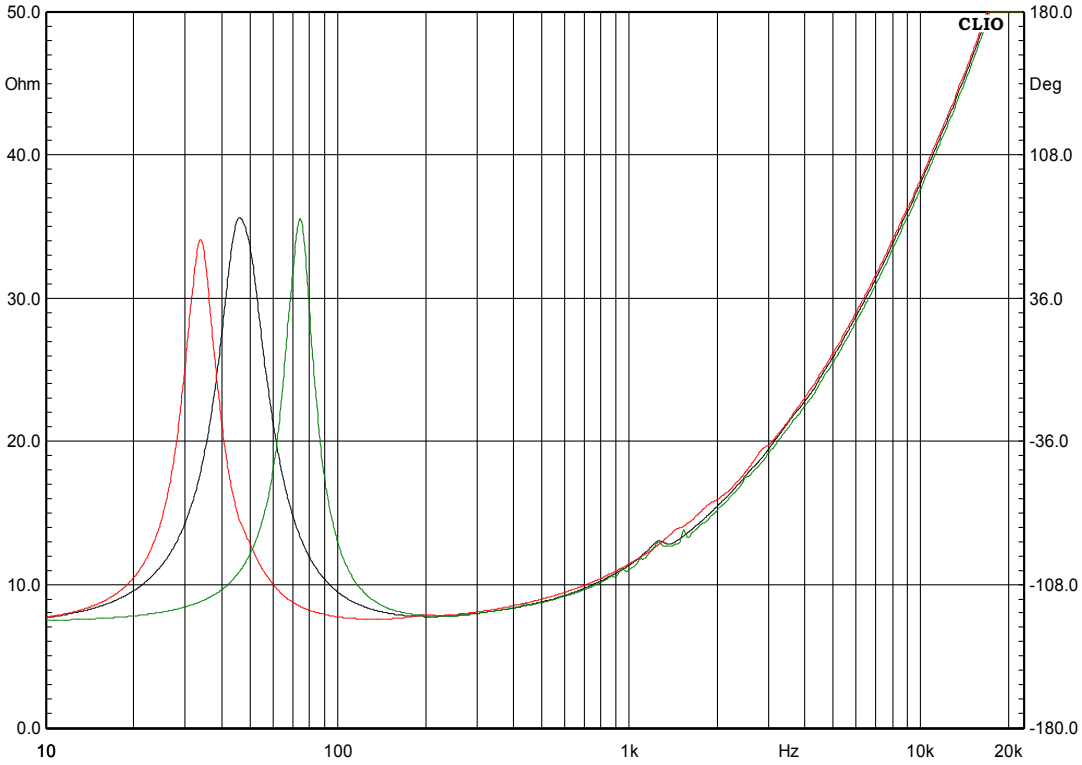
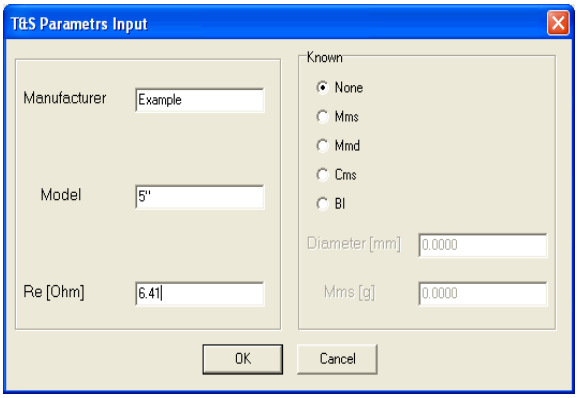
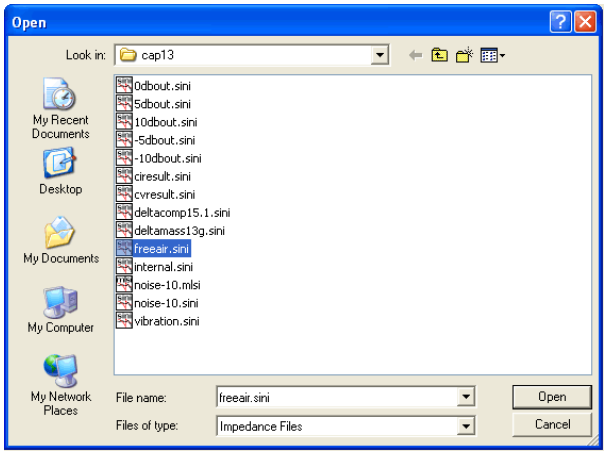


Figure 13.20

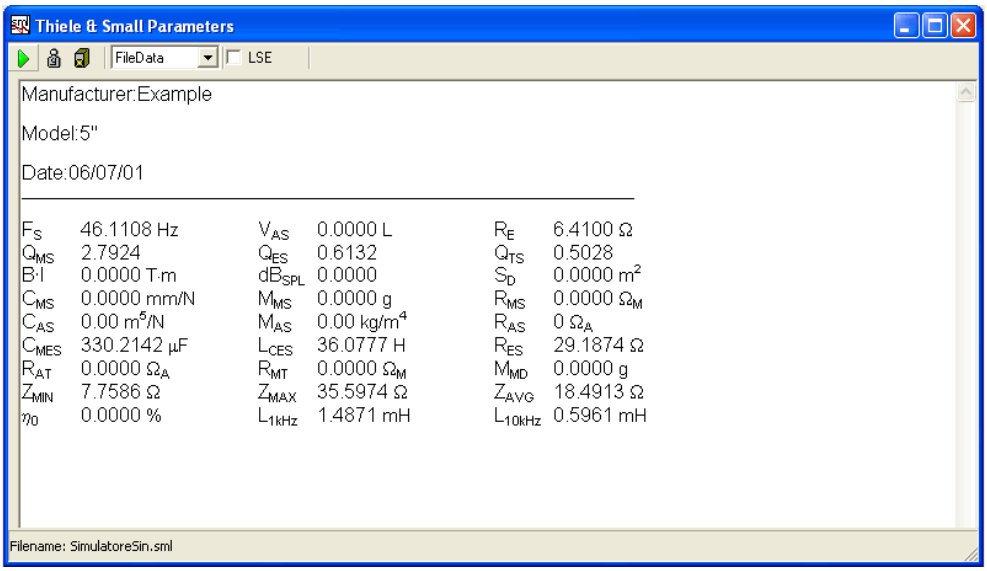
The black curve refers to free air, the red to delta mass, the green to delta compliance. As you can see adding a mass results in lowering Fs while loading the cone with a closed volume will increase Fs. Always check this is the case to avoid errors in the post processing routine. As deriving T&S parameters can be performed at any subsequent time after the impedance measurements has been taken, we suggest you save them with meaningful names as well as using the Notes Dialog to store the measuring conditions as mass weight and/or volume. The last step before pressing Go in the T&S dialog is to measure both the voice coil DC resistance and the effective cone diameter in millimeters. In this example they are 6.41 Ohm and 133mm. Finally we select File Data as the data origin and pressing Go we get the following prompt Dialog



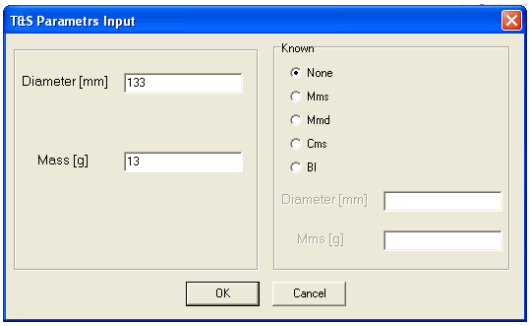
After the correct values have been typed in and clicking OK we will be prompted for the file name. The file required here is the free air impedance measurement.



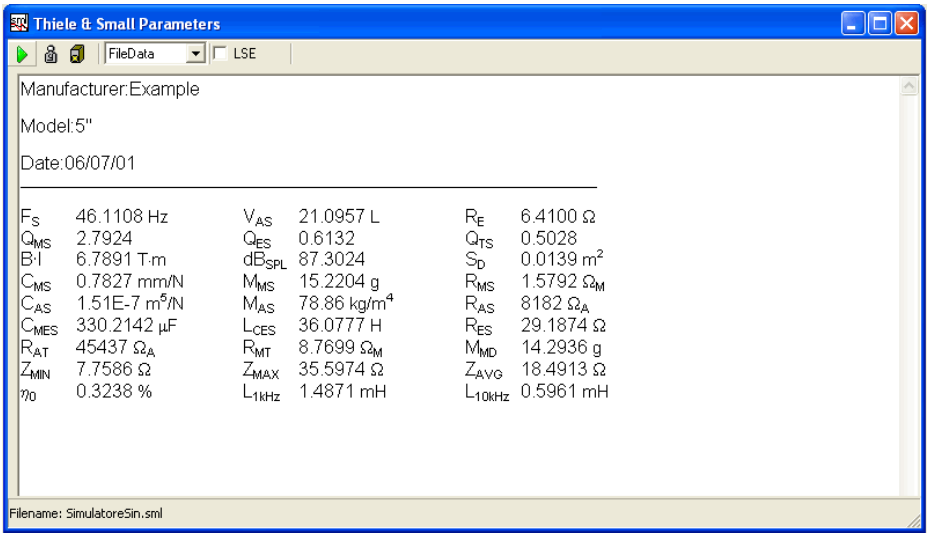
Opening the file we get this partially filled T&S parameters screen.



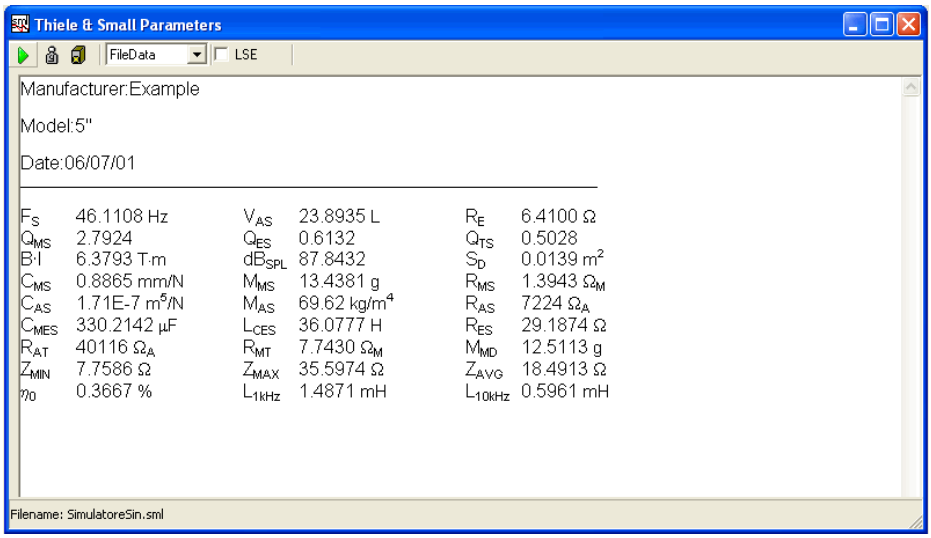
Now we can save this result for later use or proceed immediately for the missing parameters. Notice that the two Buttons for Delta Mass and Delta Compliance that were disabled before are now enabled. We Click on the Delta Mass Button and type in the required values



and, finally, after we have choose the impedance file obtained with the added mass, we get our completely filled T&S parameters screen



We can now save our complete results and proceed with the Delta Compliance. The free air derived data is already in memory and we can finally deal with the last part of the procedure, which is nearly the same as before. We will be prompted for volume instead of weight. Obviously the file we have to choose is relative to the driver loaded with a known volume (15.1 liters in this case). Here we show the results for the Delta Compliance method. The two sets of data do agree pretty well.



13.7.5 USING LSE (LEAST SQUARE ERROR)

With this option checked, T&S parameters are calculated in a more sophisticated way. Basically the parameters obtained in the standard way are used to derive the starting values for the equivalent electric circuit model. These values are then changed slightly around the initial value until the best fit between the calculated and measured impedance curve is found in a large frequency range around resonance. This procedure works fine and gives an accuracy advantage if the driver impedance fits the Thiele and Small model pretty well. Unfortunately this is not always the case. Should the user encounter such cases (if the parameters obtained with and without LSE differ substantially) the user must be aware that the entire concept of T&S parameters has lost its usefulness. This often happens with tweeters or woofers whose impedance has been measured at a high level with defective devices.

14 LINEARITY & DISTORTION

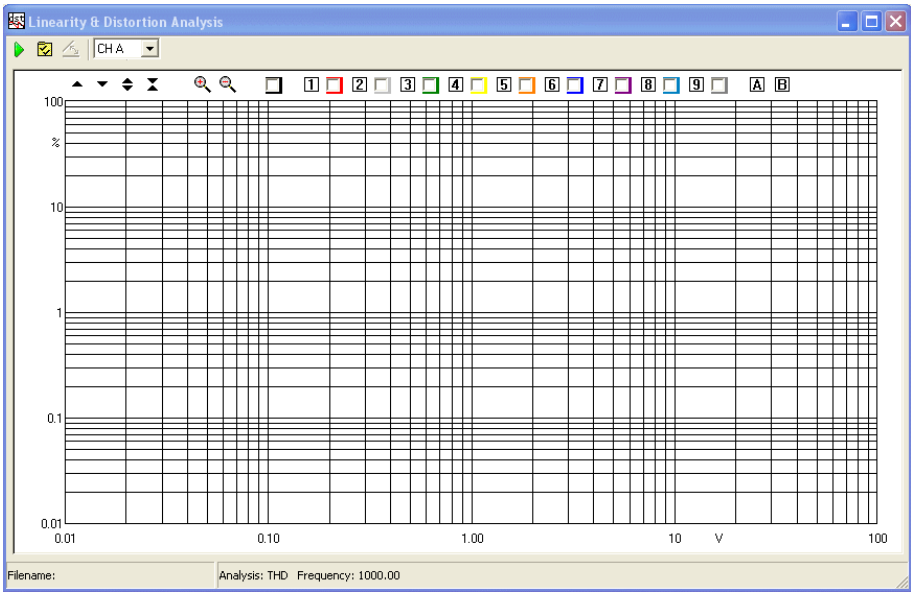
14.1 INTRODUCTION

Linearity and Distortion analysis are grouped together though they are, apparently, opposite terms.

From the analyser point of view however, they are similar as either the fundamental or the harmonics (intermodulation) amplitude is evaluated while sweeping D.U.T. input level.

Defining applications for these analysis would bring to and endless list, limited only by user fantasy; however the entire menu is oriented to perform measurements on electronic devices as power amplifiers or preamplifiers. The X axis units can be selected between Volts and Watts. So please don't put a microphone in front of a loudspeaker expecting useful results except, maybe, destroying the unit.

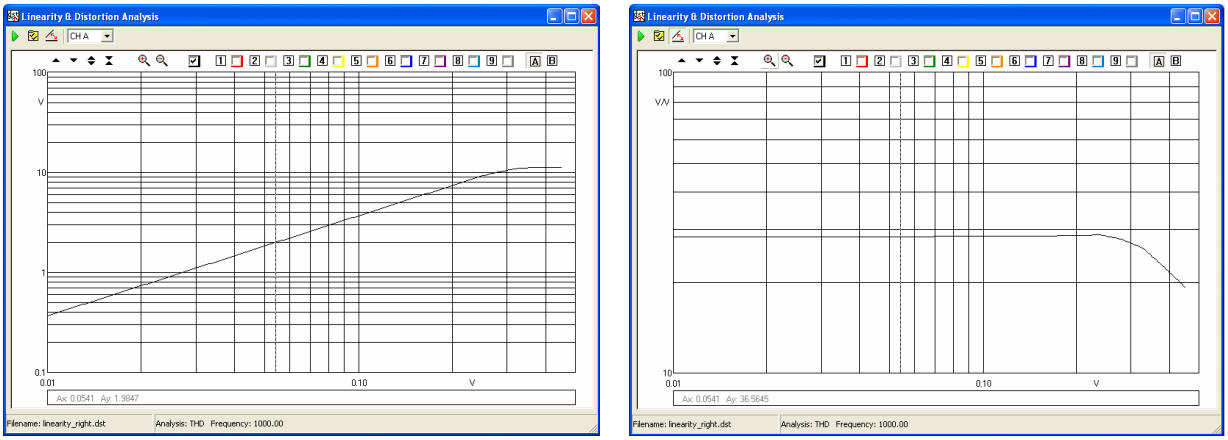
14.2 LINEARITY & DISTORTION CONTROL PANEL



Starts a Linearity and Distortion measurement.

Enters the Linearity and Distortion settings dialog box.

Compute Linearity, and is enabled only if Linearity analysis is selected (in the settings dialog, see below). Basically the process consists in dividing the measured output by the supplied input. Doing this greatly simplify realising both DUT gain and deviation from linearity. The following example should clarify things further.



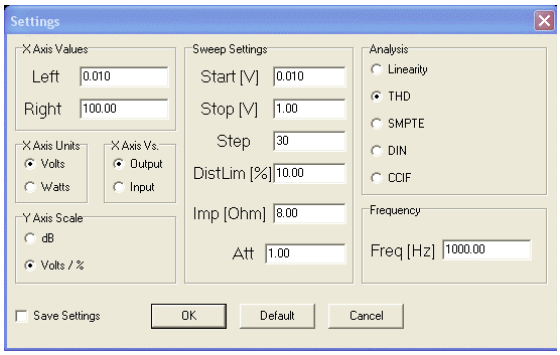
The graphs refers to a linearity measurement of a Push Pull tube amp. After processing, the Y scale can be expanded, still including the whole span, greatly enhancing detail inspection.

14.2.1 TOOLBAR DROP DOWN LIST

Input channel

Selects the input channel configuration

14.2.2 LINEARITY & DISTORTION SETTINGS DIALOG



X Axis Values

Allows setting the X axis extreme left and right values. Has immediate effect, once OK is pressed, and has only graphical implication; that is it does not affect the actual or next measure span.

X Axis Unit

Can be either Volts or Watts. Has immediate effect and the curve is recalculated accordingly. Changing the impedance in the sweep settings does not affect results as the impedance set at measuring time is taken for calculation. When Vs. input is selected in the **X Axis Vs.**, Volts is forced and the choice disabled.

X Axis Vs.

Selects if the X axis represents DUT output or input values. Input Values can only be expressed in Volts.

Y Axis Scale

Selects how the Y axis is displayed. When anything but **Linearity** is selected in the **Analysis** radio button panel, selecting **Volts/%** will display distortion in percent in a bi-logarithmic graph. Selecting dB will plot distortion in dB below

fundamental/s. When **Linearity** is selected dBV or Volts will be used as Y Unit. If the compute linearity button is pressed dB or V/V are used.

Sweep Settings

These are all settings affecting the next measure to be performed. Must be therefore handled with care.

Start and Stop

Sets the voltage sweep range supplied to the DUT input. Start should be lower in value than Stop. While these values can be chosen in an iterative way, having a rough idea of the DUT gain is a good practice. Notice however that keeping the DistLim parameter to or lower than 10% would prevent hard overload of the DUT. Sweep range stops anyway when the maximum allowed distortion is reached, whatever **Stop** value is chosen. **Stop** value cannot be grater than 3V, being this the CLIO's limit.

Step

Sets sweep resolution in logarithmic equal step.

DistLim

Sets, in percentage, the maximum allowed distortion before the sweep stop anyway.

Imp

In Ohm, displays the impedance used by the system to calculate power (use CLIO Options>Units Conversion to set, see 5.4.2). Should obviously be set to the real impedance that loads the DUT. Changing this value after the measurement has no effect on an already done measure; the value should be correct **before** the measure is taken.

Att

Sets the value of an eventually used passive attenuator placed on CLIO's input. CLIO's input accept up to 100VRMS, that is 1250W/8Ohm. Should you need more, use a resistors divider on the input and set the **Att** value accordingly. Advised value are 90-10 Ohm to divide by 10.

Analysis

Through this radio button control the analysis type is selected.

Linearity

Measures the output Voltage Vs input Voltage. Output level is measured through FFT looking only at the amplitude of the generated tone. That is harmonics noise or other is not considered in the output level.

THD

Measures Total Harmonic Distortion. Again distortion level is calculated via FFT and therefore noise is not taken in account. These, in some limited cases, at lower level, yields to slightly different results than those obtained via traditional THD+noise measurements. This approach is, however, more accurate. For setting the measurement frequency please see below.

SMPTE

Measures Intermodulation distortion using SMPTE standard. Two tones are generated in a 4:1 ratio at 60Hz and 7000Hz. Intermodulation components up to the 5^{th} order are considered for distortion.

DIN

Measures Intermodulation distortion using DIN standard. Two tones are generated in a 4:1 ratio at 250Hz and 8000Hz. Intermodulation components up to the 5^{th} order are considered for distortion.

CCIF

Measures Intermodulation distortion using two equal level near spaced (1kHz) in Frequency tones. Difference Intermodulation components up to the 2^{th} order are considered for distortion. To keep results directly comparable with THD analysis both output Voltage or Power are single tone equivalent scaled. Considering a power amplifier clipping point, this usually due to the peak value of the signal rather then it RMS value. With two tone of equal level the RMS values is 3dB lower than the same peak to peak single tone.

Freq

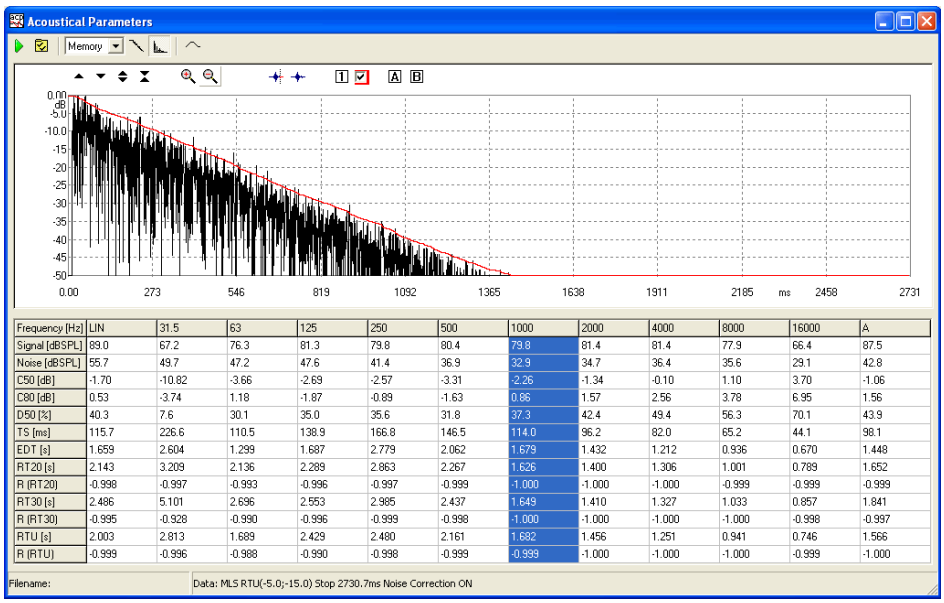
Has no effect in SMPTE and DIN analysis which use fixed frequencies. Set the measurement frequency for Linearity and THD. Set the center Frequency for CCIF; for example 15500Hz means two tone are generated, spaced by 1kHz, at 15000Hz and 16000Hz

15 ACOUSTICAL PARAMETERS

15.1 INTRODUCTION

With the Acoustical Parameters control panel it is possible to evaluate the acoustical behavior of a room and carry out sophisticated post processing of a measured impulse response to calculate the acoustical parameters as defined by the **ISO 3382** standard. These quantities describe the behavior of auditoria, concert halls and are applicable to any room intended for speech or music reproduction.

15.2 THE ACOUSTICAL PARAMETERS CONTROL PANEL



Frequency [Hz]	LIN	31.5	63	125	250	500	1000	2000	4000	8000	16000	A
Signal [dBSPL]	89.0	67.2	76.3	81.3	79.8	80.4	79.8	81.4	81.4	77.9	66.4	87.5
Noise [dBSPL]	55.7	49.7	47.2	47.6	41.4	36.9	32.9	34.7	36.4	35.6	29.1	42.8
C50 [dB]	-1.70	-10.82	-3.66	-2.69	-2.57	-3.31	-2.26	-1.34	-0.10	1.10	3.70	-1.06
C80 [dB]	0.53	-3.74	1.18	-1.87	-0.89	-1.63	0.86	1.57	2.56	3.78	6.95	1.56
D50 [%]	40.3	7.6	30.1	35.0	35.6	31.8	37.3	42.4	49.4	56.3	70.1	43.9
TS [ms]	115.7	226.6	110.5	138.9	166.8	146.5	114.0	96.2	82.0	65.2	44.1	98.1
EDT [s]	1.659	2.604	1.299	1.687	2.779	2.062	1.679	1.432	1.212	0.936	0.670	1.448
RT20 [s]	2.143	3.209	2.136	2.289	2.863	2.267	1.626	1.400	1.306	1.001	0.789	1.652
R (RT20)	-0.998	-0.997	-0.993	-0.996	-0.997	-0.999	-1.000	-1.000	-1.000	-0.999	-0.999	-0.999
RT30 [s]	2.486	5.101	2.696	2.553	2.985	2.437	1.649	1.410	1.327	1.033	0.857	1.841
R (RT30)	-0.995	-0.928	-0.990	-0.996	-0.999	-0.998	-1.000	-1.000	-1.000	-1.000	-0.998	-0.997
RTU [s]	2.003	2.813	1.689	2.429	2.480	2.161	1.682	1.456	1.251	0.941	0.746	1.566
R (RTU)	-0.999	-0.996	-0.988	-0.990	-0.998	-0.999	-0.999	-1.000	-1.000	-1.000	-0.999	-1.000

Fig. 15.1 The Acoustical Parameters control panel

In Fig. 9.1 you can see the Acoustical Parameters control panel; this figure shows an octave filtered impulse response (at 1kHz) presented both as reverberant decay and ETC; then all the calculated parameters are listed in tabular form.

The source of any acoustical parameters calculation is a measured **Impulse Response**; CLIO gives you this possibility by means of the MLS&LogChirp menu; please refer to chapter 10 where it is described how to measure the impulse response of a room using MLS or LogChirps.

For a detailed description of the graphical display (common also to other measurement control panels) and its capabilities please refer to Chapter 6. For a description of the available shortcuts please refer to section 5.5.2.

15.2.1 TOOLBAR BUTTONS AND DROP DOWN LISTS

Starts an Acoustical Parameters calculation. See below the data source for the calculation.

Enters the Acoustical Parameters Settings dialog box.

impulse response data source drop down

Selects the source of impulse response data among the following:

- **Memory**. The impulse response is already in memory and is reprocessed with current settings.
- **File**. The impulse response is loaded from disk.
- **MLS**. Current impulse response loaded in the MLS control panel is processed.

Enters the **impulse display mode** and shows the Schroeder reverberant decay relative to the selected fraction of octave. The impulse response under processing is first octave filtered and then the Schroeder decay evaluated.

Enters the impulse display mode and shows the ETC relative to the selected fraction of octave.

 Enters the **frequency display mode** and shows the selected acoustical parameter versus frequency behavior.

Selects the time instant after which the impulse response data are discarded. The parameters calculation start from this point backwards. It permits you to eliminate unwanted behaviors of the room under test and measurement artifacts.

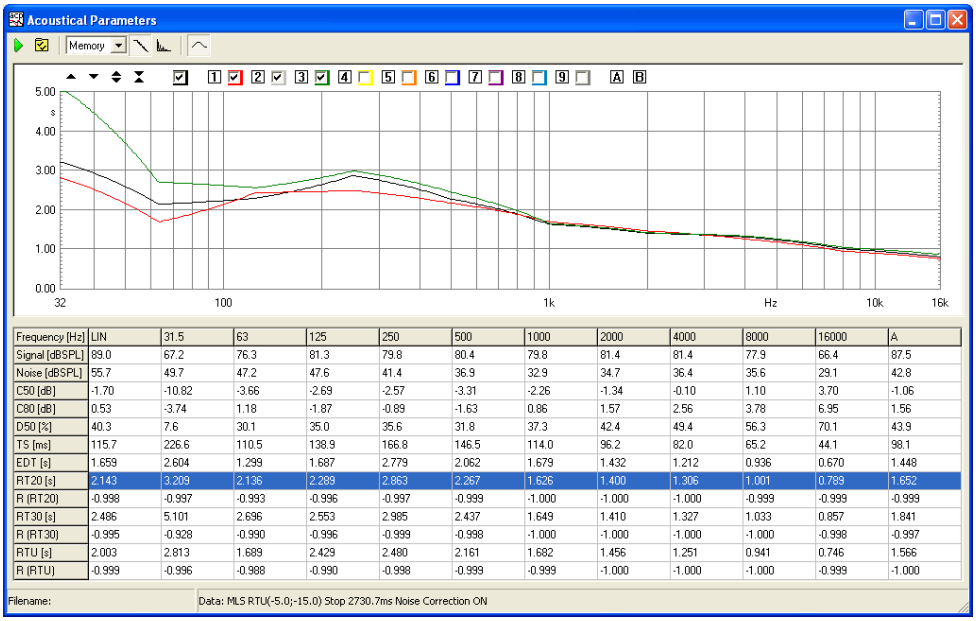
15.2.2 INTERACTION WITH THE A.P. CONTROL PANEL

It is possible to interact with the acoustical parameters control panel simply clicking on the parameters data table.

To enter the **impulse display mode** simply click on the table **first row** and select the desired octave band of interest; the selected column will change accordingly and the decay (or ETC) will also follow. The Fig.15.1 shows the selection of the 1kHz octave to which corresponds the ETC calculated.

The **leftmost column** gives wideband parameters (**Lin**) calculated over the whole available bandwidth. The **rightmost column** gives wideband parameters (**A**) calculated applying an A-Weighting filter.

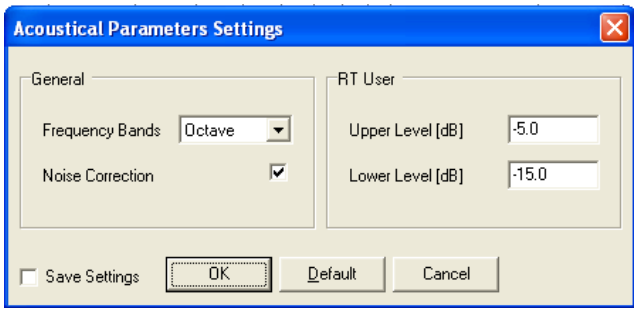
To enter the **frequency display mode** simply click on the table **first column** and select the desired parameter whose behavior versus frequency should be displayed; the selected row will change accordingly. The following figure shows the selection of the RT20 row and its behavior vs. frequency (black curve); in comparison the overlays show RT30 (green curve) and RTUser (red curve). It is not possible to selects the three rows of the correlation coefficients (see below) of the calculated RT60.



Frequency [Hz]	LIN	31.5	63	125	250	500	1000	2000	4000	8000	16000	A
Signal [dBSPL]	89.0	67.2	76.3	81.3	79.8	80.4	79.8	81.4	81.4	77.9	66.4	87.5
Noise [dBSPL]	55.7	49.7	47.2	47.6	41.4	36.9	32.9	34.7	36.4	35.6	29.1	42.8
C50 [dB]	-1.70	-10.82	-3.66	-2.69	-2.57	-3.31	-2.26	-1.34	-0.10	1.10	3.70	-1.06
C80 [dB]	0.53	-3.74	1.18	-1.87	-0.89	-1.63	0.86	1.57	2.56	3.78	6.95	1.56
D50 [%]	40.3	7.6	30.1	35.0	35.6	31.8	37.3	42.4	49.4	56.3	70.1	43.9
TS [ms]	115.7	226.6	110.5	138.9	166.8	146.5	114.0	96.2	82.0	65.2	44.1	98.1
EDT [s]	1.659	2.604	1.299	1.687	2.779	2.062	1.679	1.432	1.212	0.936	0.670	1.448
RT20 [s]	2.143	3.209	2.136	2.289	2.863	2.267	1.626	1.400	1.306	1.001	0.789	1.652
R (RT20)	-0.998	-0.997	-0.993	-0.996	-0.997	-0.999	-1.000	-1.000	-1.000	-0.999	-0.999	-0.999
RT30 [s]	2.486	5.101	2.696	2.553	2.985	2.437	1.649	1.410	1.327	1.033	0.857	1.841
R (RT30)	-0.995	-0.928	-0.990	-0.996	-0.999	-0.998	-1.000	-1.000	-1.000	-1.000	-0.998	-0.997
RTU [s]	2.003	2.813	1.689	2.429	2.480	2.161	1.682	1.456	1.251	0.941	0.746	1.566
R (RTU)	-0.999	-0.996	-0.988	-0.990	-0.998	-0.999	-0.999	-1.000	-1.000	-1.000	-0.999	-1.000

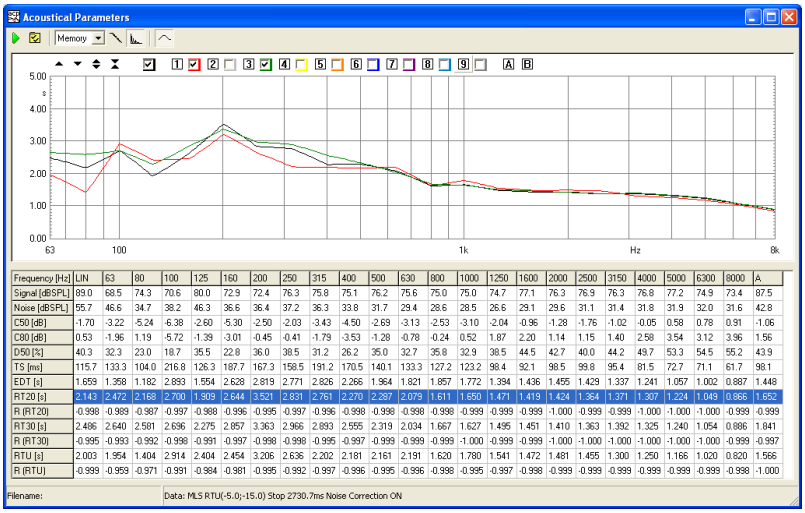
Filename:
Data: MLS RTU(-5.0;-15.0) Stop 2730.7ms Noise Correction ON

15.3 ACOUSTICAL PARAMETERS SETTINGS



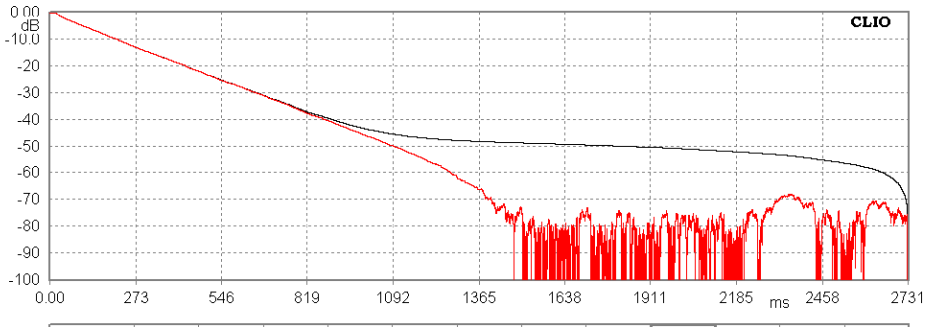
Frequency Bands

Selects either Octave or Third of Octave calculations. The following figure shows the same data analyzed before in octave bands now presented with 1/3 of octave processing.



Noise Correction

Applies noise correction to the tail of the impulse response as suggested by ISO 3382. The figure below shows the increase in the linear portion of the calculated decay which is obtainable.



RT User

It is possible to input the upper and lower level, in dB, used for RTUser calculations.

15.4 THE CALCULATED ACOUSTICAL PARAMETERS

The acoustical parameters are calculated from a measured **decay curve**. A decay curve is defined as the decay of sound pressure level as a function of time after the sound source has ceased. The decay curves are calculated from the measured impulse response after octave filtering has been applied; also wideband (linear or A-weighted) decay curves are available.

Historically the most important acoustical parameter is the **Reverberation Time** (T or RT) defined as the time, in seconds, that would be required for the sound pressure to decrease by 60dB after the sound source has ceased; this is why the most used indication you find in literature for the reverberation time is **RT60**. As it is normally practically difficult to obtain a sufficient dynamic range to measure the reverberation time **directly**, the norm provides for its evaluation based on a smaller decay of 30dB; in this case the reverberation time, indicated as RT30, should be the time evaluated considering a linear decay given by the least-square regression of the measured curve from -5dB to -35dB. Also provided is the possibility of evaluating RT20 and RTUser (based on user defined limits).

Sound level parameters.

Signal [dBSPL]. Measured signal level in dBSPL in the band of interest.

Noise [dBSPL]. Measured background noise level in dBSPL in the band of interest.

Balance between early and late arriving energy.

C50 [dB]. 50ms early-to-late arriving sound energy ratio (i.e. ratio between the energy arrived in the first 50 milliseconds to the energy arrived after). C50 is usually evaluated when results relate to speech reproduction.

C80 [dB]. Usually named "Clarity". 80ms early-to-late arriving sound energy ratio. C80 is usually evaluated when results relate to music reproduction.

D50 [%]. Usually named "Definition". Directly relates to C50 with the following equation:

$$C_{50} = 10\log(\frac{D_{50}}{1 - D_{50}})dB$$

TS [ms]. Time of centre gravity of the squared impulse response. It is another measure of acoustic clarity; the higher Ts the poorer is clarity.

Decay time measurements.

EDT [s]. Early Decay Time i.e. time required to sound to decrease of 10dB from the initial maximum level. EDT is directly related to the perceived reverberation while reverberation time relates to the physical properties of the room.

RT20 [s]. Reverberation time evaluated from a 20 dB dynamic range (-5dB,-25dB). See also below the **correlation coefficient R** associated with RT20.

RT30 [s]. Reverberation time evaluated from a 30 dB dynamic range (-5dB,-35dB). See also below the **correlation coefficient R** associated with RT30.

RTU [s]. Reverberation time evaluated from a user defined dynamic range; refer to 15.3 acoustical parameters settings. See also below the **correlation coefficient R** associated with RTUser.

R(RT). Each reverberation time estimation (RT20, RT30 and RTU) has associated a negative number which is the correlation coefficient R showing how closely the corresponding decay curve fits a straight line. A value of -1 gives a perfect linear fit. **When the correlation coefficient is smaller than -0.95 the RT value should be viewed with suspect as the decay curve may not be sufficiently linear; direct inspection of the decay curve with markers should be carried out.**

15.5 NOTES ABOUT ACOUSTICAL PARAMETERS MEASUREMENT

This paragraph gives some guidelines that should be followed while executing the measurement of the impulse response of rooms; what said here should be considered together the general procedure that is described in chapter 10 for executing MLS measurements.

The sound source shall be as omni-directional as possible. Maximum acceptable deviation from omni-directionality should not be higher than ±1dB up to 500Hz, ±3dB at 1kHz, ±5dB at 2kHz, ±6dB at 4kHz when excited with octave bands noise and measured in a free field.

Regarding **measurement positions** it is important to execute an adequate number of measurements with different source and receiver positions to characterize the entire room. For large auditoria a number of measurements from 6 to 10 in dependance of the number of seats (from 500 to 2000) should be carried out.

The microphone should be placed at a height of 1.2m above the floor at audience seat locations to be representative of listener’s ear height.

15.6 STI CALCULATION

The Speech Intelligibility Index are calculated from a single measured **MLS response**.

Some care should be followed while executing the measurement of the impulse response to be used for the STI calculation:
- the procedure is valid only for an MLS stimulus as the signal to noise ratio is collected in a single measurement.
- the impulse length must be at least 1.6 seconds to correctly calculate the lowest modulation frequency needed for the MTF matrix.

The STI is a single number index that takes into account different effects that are decreasing the speech intelligibility in a room as the background noise and the reverberation. the index is calculated starting from a set of MTF (modulation transfer functions) calculated for 7 octave bands and 14 modulation frequencies as defined in the **ISO IEC 60268-16:2003** standard. The 7 x 14 matrix is then reduced to a vector of 7 MTI modulation transfer indexes and combined into a single number index.

The text file is divided in three sections:

STI index- "classic" STI index calculation as defined in IEC 60268-16:1998. The STI index is rated accordingly to the following table:

0	<	STI	<	0.3	BAD
0.3	<	STI	<	0.45	POOR
0.45	<	STI	<	0.6	FAIR
0.6	<	STI	<	0.75	GOOD
0.75	<	STI	<	1	EXCELLENT

STIr index - "revised" STI index calculation as defined in IEC 60268-16:2003, two different MTI weighting are present hence two STI indexes: STImale and STIfemale.

RaSTI index - Rapid Speech Transmission index, is a simpler version of the STI calculated from only two frequency bands and 9 modulation frequencies

The parameters are calculated together with the acoustical parameters and can be viewed in text format by pressing the STI button.

```
-----------------------------------------------------------------------
STI index
-----------------------------------------------------------------------
Oct.Band  125       250       500       1k        2k        4k        8k
f1=0.63   0.716     0.776     0.726     0.781     0.794     0.842     0.933
f2=0.80   0.669     0.718     0.666     0.727     0.733     0.777     0.865
f3=1.00   0.627     0.665     0.612     0.682     0.677     0.717     0.803
f4=1.25   0.584     0.611     0.561     0.640     0.622     0.658     0.740
f5=1.60   0.530     0.545     0.514     0.597     0.561     0.590     0.670
f6=2.00   0.479     0.480     0.486     0.559     0.503     0.523     0.605
f7=2.50   0.422     0.412     0.462     0.528     0.439     0.448     0.535
f8=3.15   0.340     0.334     0.433     0.502     0.357     0.347     0.457
f9=4.00   0.297     0.302     0.428     0.456     0.238     0.174     0.375
f10=5.00  0.356     0.253     0.421     0.404     0.209     0.112     0.364
f11=6.30  0.449     0.171     0.359     0.418     0.324     0.374     0.470
f12=8.00  0.569     0.491     0.446     0.494     0.449     0.546     0.645
f13=10.00 0.407     0.398     0.379     0.494     0.406     0.508     0.690
f14=12.50 0.372     0.298     0.340     0.475     0.267     0.375     0.502
-----------------------------------------------------------------------
MTI             0.487     0.461     0.488     0.554     0.470     0.499     0.618

STI=0.509 rated Fair
ALcons=10.8%
-----------------------------------------------------------------------
STIr index
-----------------------------------------------------------------------
Oct.Band  125       250       500       1k        2k        4k        8k
f1=0.63   0.716     0.686     0.703     0.747     0.754     0.727     0.757
f2=0.80   0.669     0.648     0.649     0.702     0.704     0.691     0.731
f3=1.00   0.627     0.610     0.598     0.662     0.656     0.652     0.700
f4=1.25   0.584     0.567     0.550     0.623     0.606     0.607     0.662
f5=1.60   0.530     0.511     0.505     0.583     0.548     0.552     0.613
f6=2.00   0.479     0.452     0.478     0.548     0.492     0.494     0.561
f7=2.50   0.422     0.389     0.454     0.518     0.430     0.424     0.502
f8=3.15   0.340     0.314     0.426     0.493     0.350     0.328     0.431
f9=4.00   0.297     0.283     0.421     0.448     0.232     0.158     0.353
f10=5.00  0.356     0.235     0.414     0.397     0.203     0.097     0.342
f11=6.30  0.449     0.154     0.353     0.411     0.318     0.354     0.443
f12=8.00  0.569     0.462     0.439     0.485     0.440     0.514     0.594
f13=10.00 0.407     0.376     0.373     0.485     0.398     0.480     0.628
f14=12.50 0.372     0.279     0.334     0.466     0.261     0.355     0.473
Lk [dB]         88.1      84.9      84.7      91.0      93.6      94.7      95.8
-----------------------------------------------------------------------
MTI             0.487     0.426     0.478     0.541     0.457     0.459     0.557
```

```
STIr(male)=0.490   rated Fair
ALcons=12.0%
STIr(female)=0.487 rated Fair
ALcons=12.2%
-------------------------------------------------------------------------
RaSTI index
-------------------------------------------------------------------------
Oct.Band      500  2k
0.7                     0.766
1.0                0.612
1.4                     0.594
2.0                0.486
2.8                     0.402
4.0                0.428
5.6                     0.260
8.0                0.446
11.2                    0.360
-------------------------------------------------------------------------

RaSTI=0.484   rated Fair
ALcons=12.4%
```

The **ALcons** index - Articulation Loss of Consonants - is directly related to the STI by means of a mathematical formula.

16 Leq LEVEL ANALYSIS

16.1 INTRODUCTION

With the Leq Analysis control panel it is possible to execute **real-time capture and level measurement** of any kind of signal present at CLIO's input. The behavior of the instrument closely resemble that of a graphical level recorder plus direct-to-disk data capture.

When analyzing an acoustical event this control panel gives you complete information about the equivalent continuous sound level (Leq) and related quantities according to IEC 61672 standard; if used together the FFT frequency analysis you get a complete **integrating sound level meter**.

16.2 THE Leq CONTROL PANEL

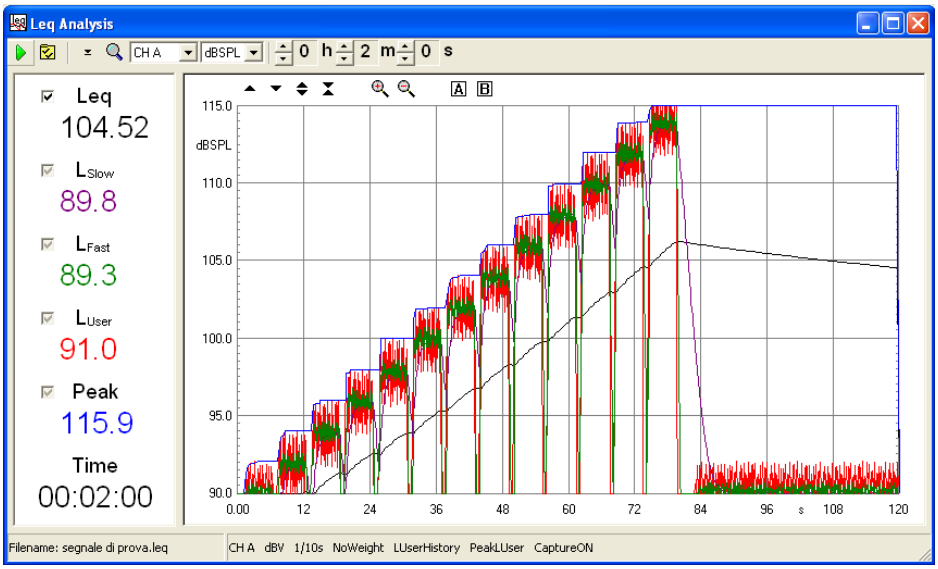


Fig. 16.1 The Leq control panel

In Fig. 16.1 you can see the Leq Analysis control panel; this figure shows a low frequency signal, increased in 2dB steps each maintained for 6 seconds, used for subwoofers power handling capability. To be noted the **equivalent level Leq** (black curve), the **time history** (red curve), the **peak level** (blue curve) and the levels with **slow and fast** integration (purple and green curves).

For a description of the available shortcuts please refer to section 4.5.2.

16.2.1 TOOLBAR BUTTONS AND DROP DOWN LISTS

Starts a Leq acquisition and analysis. If data capture is active the event is automatically registered on the hard disk.

Invokes an FFT measurement together the Leq one.

Enters the Leq Analysis Settings dialog box.

When pressed, resets peak value. Does not affect any other calculation.

Activates real time data display; useful for high resolution time measurements (1/100s and 1/1000s).

channel display

Selects the input channel to display among the following:

- Channel A only
- Channel B only

Y scale units

Selects the measurement units among the following:

- dBV
- dBu
- dBRel (with respect to the global reference level; see chapter 8)
- dBSPL (for acoustical measurements)
- dBPa (for acoustical measurements)
- dBmeter (for laser measurements)
- dBSm/s (for laser measurements)
- dBSm/s^2 (for acceleration measurements)

Stop Hours, Minutes and Seconds

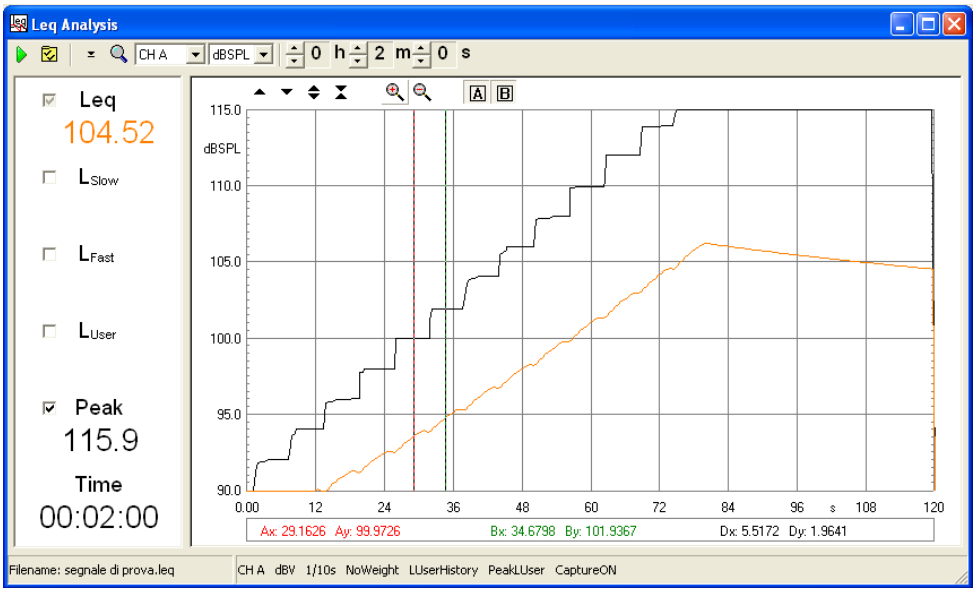
Inputs the measurement stop time. It is possible to select up to 23h59m59s measurement time.

16.2.2 INTERACTION WITH THE Leq CONTROL PANEL

It is possible to interact with the Leq control panel clicking on the left data display where you can find five **three state checkboxes**.

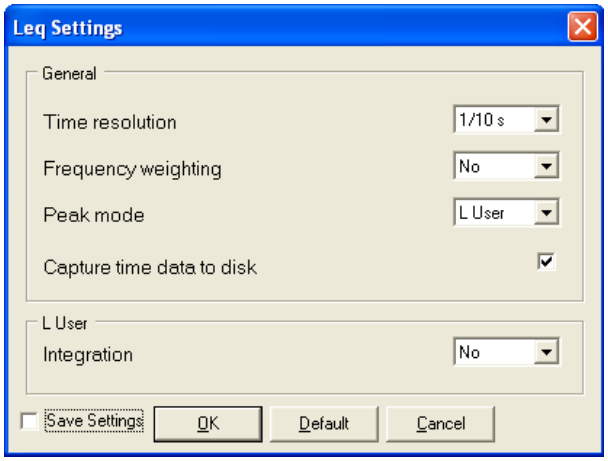
Each checkbox refers to one calculation and data curve. Its state can be:
Deselected. The data value and corresponding curve are **NOT** displayed.
Selected. The data value and corresponding curve are displayed with their color.
Active. The data value and corresponding curve are displayed in black and curve values are inspectable with the A and B markers.

In the following figure you may see the same measurement presented in figure 16.1; we have done the following; the fast, slow and time history levels have been deselected, i.e. hidden, the peak level is now active (black curve) and inspected with the two markers while the Leq is simply visible (orange curve and value).



Below the five calculated data value you can see the actual time display which changes during measurement or, at the end, states the total duration of it.

16.3 Leq SETTINGS



Time resolution

Selects the time resolution of the measurement. It is possible to choose a value among 1s, 1/2s, 1/4s, 1/10s, 1/100s and 1/1000s. Normally choose the least resolution possible as this choice directly reflects on the measured data size (**.leq** binary files). This setting is not influencing the sampling frequency that remains 48000Hz.

Frequency weighting

Selects the frequency weighting applied; you can choose either No Weight or A-Weighting.

Peak mode

Selects how the peak is measured. You may select between the two:

- **Max**. The peak is the maximum value acquired given the particular time resolution and frequency weighting.
- **LUser**.The peak is the maximum value of the LUser measurement.

Capture time data to disk

If active, during the measurement the acquired time data is saved to the hard disk. It is then possible to create a standard wave file of the event measured for later post processing. Be extremely careful when activating this feature as it requires a huge amount of disk space: circa **6MB/min** or **0.35GB/hour**.
Maximum recordable time is 12 hours.

LUser integration

Selects the time integration of the user definable level measurement. You may choose among the following:

- **No**. No integration is applied; the result is that the classical time history is displayed.
- **Impulse**. Classical Impulse integration, 35ms time constant with 2.9dB/s decay rate.
- **35ms**. Modified impulse integration; only 35ms time constant.

17 WOW AND FLUTTER

17.1 INTRODUCTION

Within this menu Wow & Flutter measurements are possible, meeting both IEC and NAB standards. Basically, what is measured is the frequency modulation that follows instantaneous speed variations due to mechanical imperfections in analog recording or playback devices. Differently than in traditional Wow & Flutter analyser the whole measuring process is taken digitally without relying on analog FM discriminator, filter and detector with much higher accuracy, limited by the clock quartz only. Furthermore, aside traditional number data, a time graph is presented as well as a Frequency Domain Analysis of the demodulated signal. The latter feature greatly simplify locating the cause of problems, once rotating speed and mechanical circumference of eventually defective rotating parts are known. Carrier frequency can range form 1500Hz to 6000Hz. This is important if you record directly the test signal. Using existing test support, IEC specify a test frequency of 3150Hz, NAB of 3000Hz.

17.2 WOW & FLUTTER CONTROL PANEL

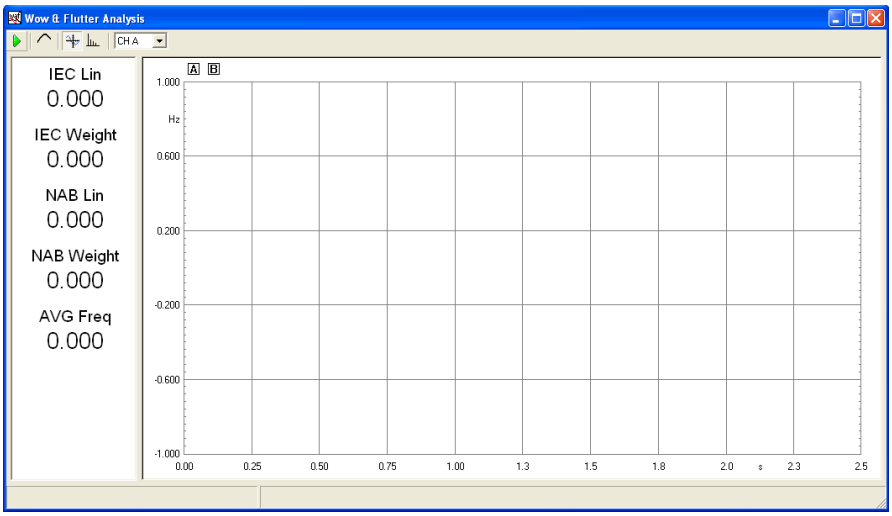


Figure 17.1

17.2.1 TOOLBAR BUTTON

Starts a Wow & Flutter analysis.

Enables weighting filter in the time domain windows. Frequency Domain remains unaffected. Works both when the analysis is running or as a post process.

Switch to Time Domain.

Switch to Frequency Domain.

17.2.2 TOOLBAR DROP DOWN LIST

Input channel

Selects the input channel configuration

17.3 FEATURES

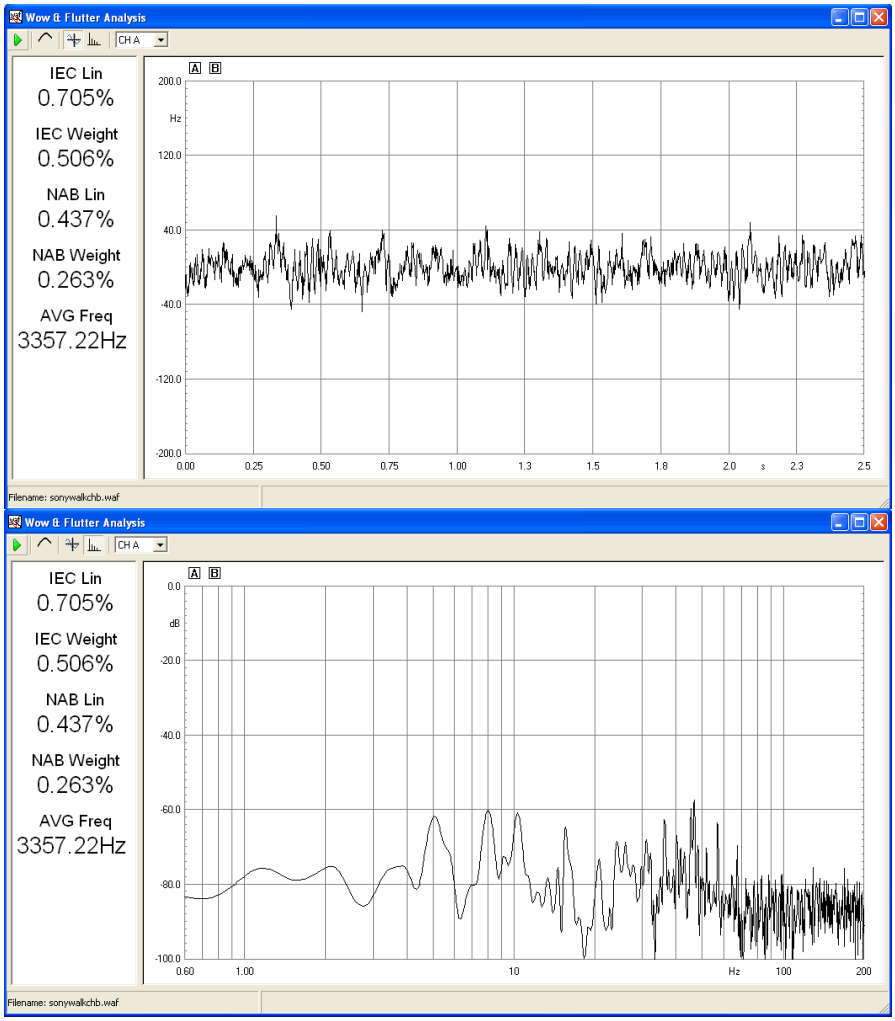


Figure 17.2

Aside a self explaining graphical part, on the left part several numeric data are present simultaneously. From top to bottom they are:

IEC LIN

expressed in percentage, express the WOW & FLUTTER value, unweighted, following IEC standard.

IEC WEIGHT

expressed in percentage, express the WOW & FLUTTER value, weighted, following IEC standard.

NAB LIN

expressed in percentage, express the WOW & FLUTTER value, unweighted, following NAB standard.

NAB WEIGHT

expressed in percentage, express the WOW & FLUTTER value, weighted, following NAB standard.

AVG Freq

expressed in Hertz is the frequency of the carrier tone. Is a direct indication of a static speed error.

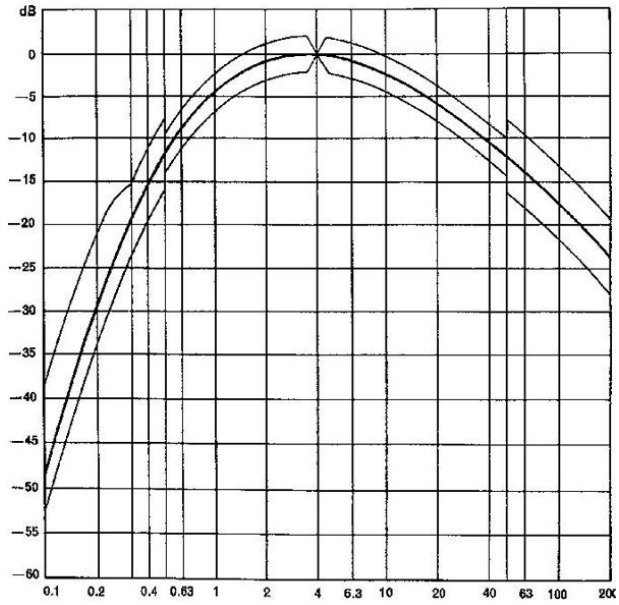


Figure 17.3

In the above figure the weighting filter response is displayed. This apply both to IEC and NAB standards. Aside carrier Frequency the main difference between them is the detector that evaluate the demodulated signal, which is peak detection in IEC and RMS in NAB; IEC Wow & Flutter values are usually greater.

18 WAVELET ANALYSIS

18.1 INTRODUCTION

The Wavelet Analysis tool allows to post-process impulse responses and to create color plots of the energy of the signal versus time and frequency. The tool is similar to the ETF analysis described in chapter 12, but since it is based on wavelet transform instead of Fourier Transform, does not suffer from the fixed time-frequency resolution.

The ETF analysis is based on Short Time Fourier Transform (STFT). The idea behind STFT is to show the temporal evolution of the signal by means of the division of the signal itself into short sections and then Fourier Transform every section. In this way the joint time-frequency evolution of the signal is highlighted. But at the same time the process lead to a fixed time and frequency resolution, due to the fact that time resolution is linked to section duration and frequency resolution is linked to FFT size.

The Wavelet Analysis tool implemented in CLIO uses a kernel of modified complex Morlet wavelets and can be interpreted as a constant Q analysis. Time resolution is high at high frequencies and frequency resolution is not too rough at low frequencies. This kind of analysis it is particularly suited for the inspection of wideband non stationary signals as the impulse responses of loudspeakers and rooms.

As a result of the Wavelet Analysis post-processing tool a matrix of coefficients is calculated. The magnitude squared of the coefficients is directly proportional to the energy of the signal in a domain located around a certain time and frequency. The magnitude squared of the Wavelet coefficients is depicted into a color plot called Scalogram:

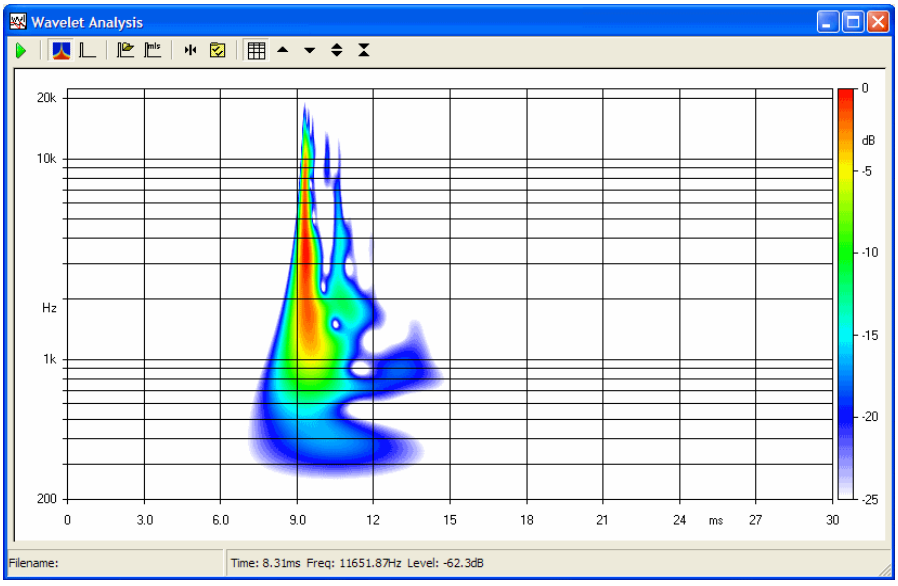


Figure 18.1 - Wavelet Analysis panel

18.2 WAVELET ANALYSIS CONTROL PANEL

Fig 18.1 show the Wavelet Analysis control panel, the behavior of this menu is similar to the Waterfall menu as seen in chapter 18.

As already stated the source of data for Wavelet Analysis is an impulse response, please refer to chapter 10 (MLS&LogChirp) to have details on how to measure an impulse response.

18.2.1 COMMON TOOLBAR BUTTONS AND DROP DOWN LISTS

Starts a Wavelet Analysis calculation.

If pressed each frequency slice of the spectrogram will be referenced to the its energy time maximum; the plot can be interpreted as the energy decay of the system.

Enters the settings dialog. See 18.3.

Moves the plot up.

Moves the plot down.

Expands the plot changing its Z scale. The Z range is reduced.

Compresses the plot changing its Z scale. The Z range is increased.

When pressed the wavelet analysis (scalogram) plot is displayed.

When pressed the grid is displayed.

When pressed the impulse response loaded in memory is displayed.

Loads an impulse response from disk (from MLS binary files).

Takes current MLS impulse response for wavelet calculation.

18.3 WAVELET ANALYSIS SETTINGS

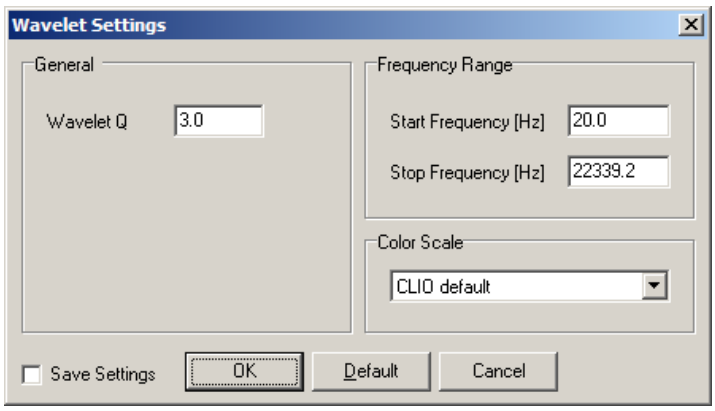


Figure 18.2 - Wavelet Settings Dialog

Start Frequency

Selects the start frequency for the analysis.

Stop Frequency

Selects the stop frequency for the analysis.

Wavelet Q

Selects the frequency resolution for the analysis, see also 18.4.1 (Q>3)

Color Scale

Selects the color gradient scale to be used:
CLIO Default - standard CLIO color map
Jet Colormap - red to blue color map

18.4 WAVELET ANALYSIS OPERATION

As already stated the data source for the Wavelet Analysis is a measured impulse response.

Once you have loaded an impulse response inside the Waterfall Analysis control panel you may easily inspect it, in the same way you also do with the MLS Impulse control panel (See chapter 10). The limits in time of the Wavelet Analysis plot will be the same of the impulse plot view. It is possible to window the impulse response (by means of a rectangular window) as in MLS&LogChirp control panel prior to the Wavelet Analysis computation.

Pressing the Go button run the Wavelet Analysis, the calculation time is dependent from the impulse response size, in older PCs and in case of very long impulse responses this can take up to 2 minutes. Computation time in a modern PC is limited to few seconds.

Once the Wavelet is calculated it is possible to zoom in/out into time without running again the Wavelet Analysis.

18.4.1 TRADING BANDWIDTH AND TIME RESOLUTION

As said before, changing the Wavelet Q parameter is it possible to trade time resolution with bandwidth resolution.

The product of temporal uncertainty and bandwidth uncertainty is fixed, the Q parameters lets adjust the frequency resolution that in turns influences time resolution.

In the following figures some examples of the same impulse response with different Q are reported.

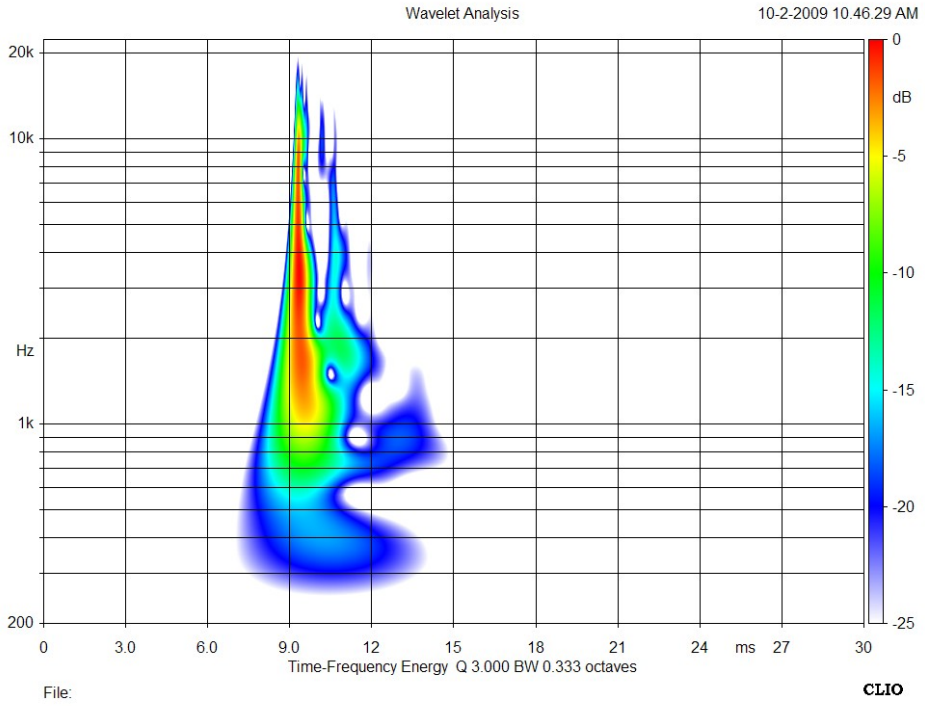


Figure 18.3 - Wavelet Analyisis of loudspeaker impulse response Q=3

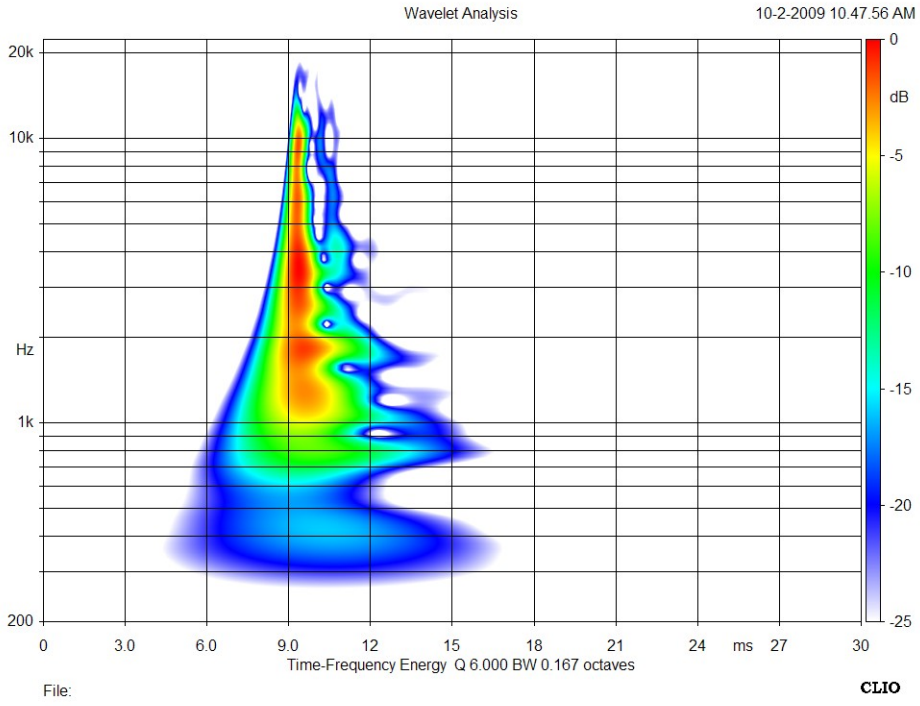


Figure 18.4 Wavelet Analysis of loudspeaker impulse response Q=6

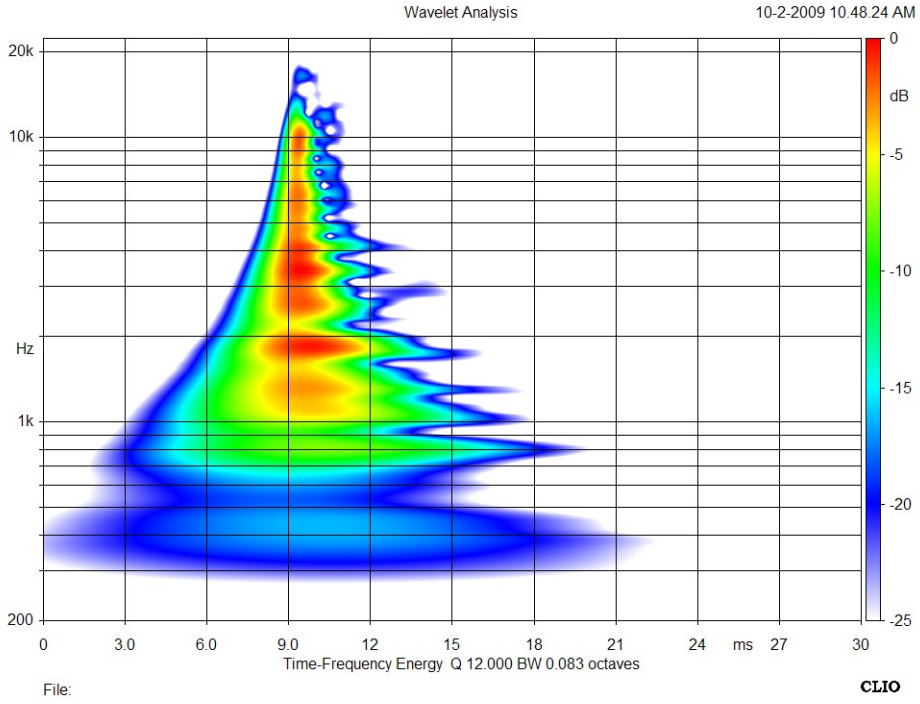


Figure 18.5 – Wavelet Analysis of loudspeaker impulse response Q=12

18.4.2 NORMALIZED SCALOGRAMS

The Scalogram is a colormap display of the magnitude square of the matrix of Wavelet coefficients. It is possible to interpret every cell of the Scalogram as proportional to the energy of the signal in a domain located around give time and frequency points.

Due to the uncertainty in time, the energy content it is smeared in time and somewhat difficult to interpret.

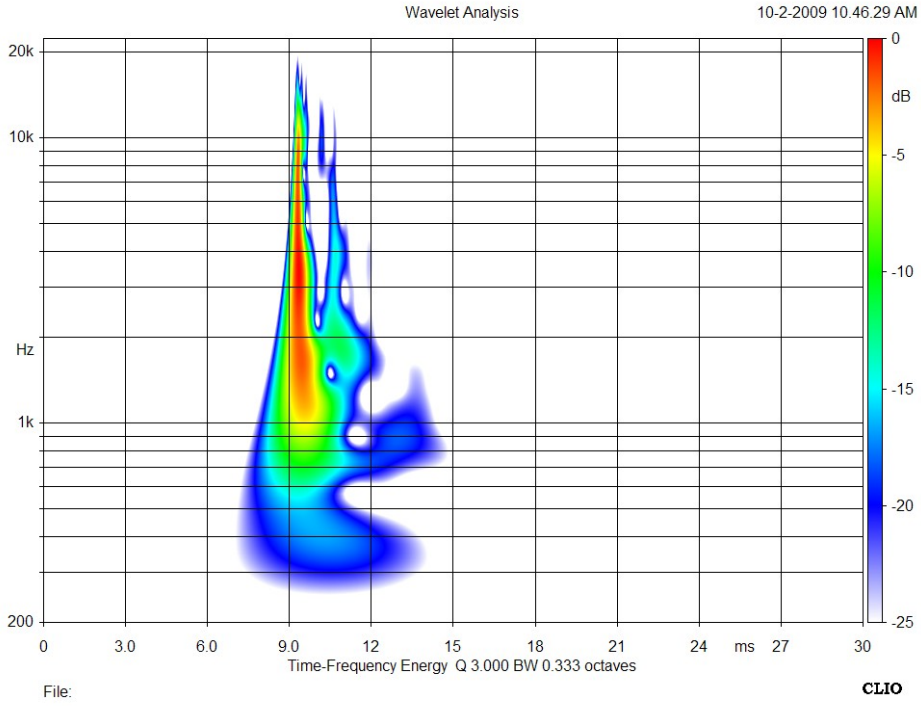


Figure 18.6 - Wavelet Analyisis of loudspeaker impulse response - Not normalized

If we are interested to the time-frequency energy decay of the system, the normalized view is more clear. Since every frequency slice it is normalized to its energy time maximum, the map can be loosely interpreted as the time-frequency energy decay of the system. The red area is a clear view of the time response of the system versus frequency and can be taught as an extension of the excess phase group delay method shown on section 10.4.4.

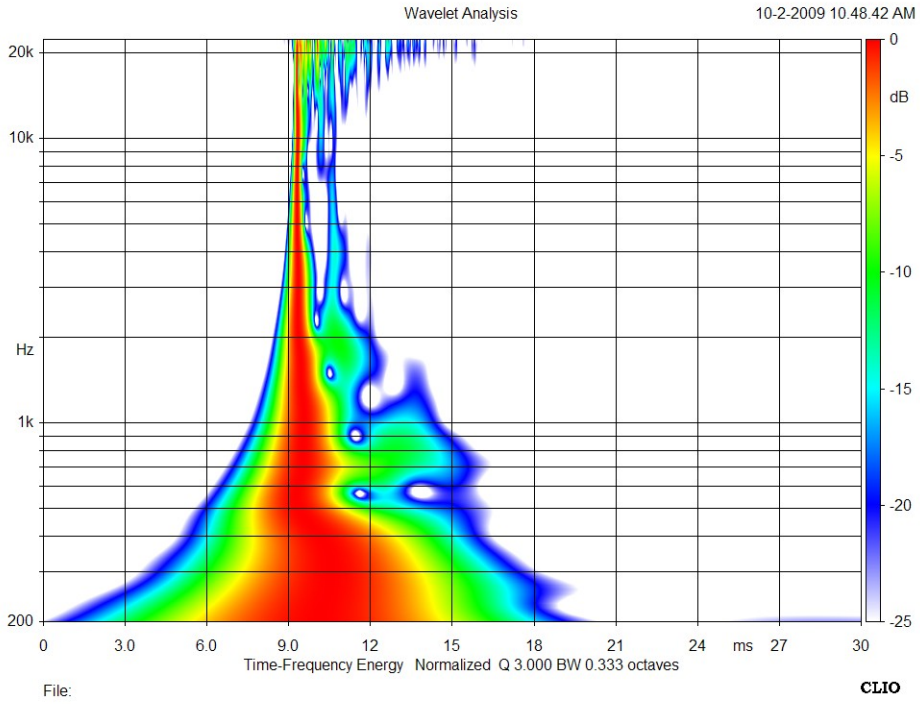


Figure 18.7 – Wavelet Analysis of loudspeaker impulse response - Normalized

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